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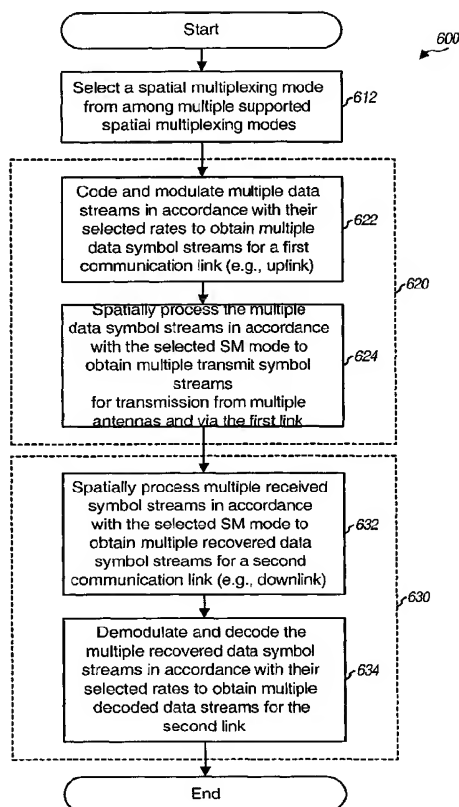
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(54) Title: MULTI-MODE TERMINAL IN A WIRELESS MIMO SYSTEM



(57) Abstract: A user terminal supports multiple spatial multiplexing (SM) modes such as a steered mode and a non-steered mode. For data transmission, multiple data streams are coded and modulated in accordance with their selected rates to obtain multiple data symbol streams. These streams are then spatially processed in accordance with a selected SM mode (e.g., with a matrix of steering vectors for the steered mode and with the identity matrix for the non-steered mode) to obtain multiple transmit symbol streams for transmission from multiple antennas. For data reception, multiple received symbol streams are spatially processed in accordance with the selected SM mode (e.g., with a matrix of eigenvectors for the steered mode and with a spatial filter matrix for the non-steered mode) to obtain multiple recovered data symbol streams. These streams are demodulated and decoded in accordance with their selected rates to obtain multiple decoded data streams.

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MULTI-MODE TERMINAL IN A WIRELESS MIMO SYSTEM

Claim of Priority under 35 U.S.C. §119

[0001] The present Application for Patent claims priority to Provisional Application Serial No. 60/421,309, entitled "MIMO WLAN System," filed October 25, 2002, assigned to the assignee hereof, and expressly incorporated by reference herein.

BACKGROUND

Field

[0002] The present invention relates generally to communication, and more specifically to a user terminal in a multiple-input multiple-output (MIMO) communication system.

Background

[0003] A MIMO system employs multiple (N_T) transmit antennas and multiple (N_R) receive antennas for data transmission and is denoted as an (N_T, N_R) system. A MIMO channel formed by the N_T transmit and N_R receive antennas may be decomposed into N_S spatial channels, where $N_S \leq \min \{N_T, N_R\}$. The N_S spatial channels may be used to transmit N_S independent data streams to achieve greater overall throughput. In general, spatial processing may or may not be performed at a transmitter and is normally performed at a receiver to simultaneously transmit and recover multiple data streams.

[0004] A conventional MIMO system typically uses a specific transmission scheme to simultaneously transmit multiple data streams. This transmission scheme may be selected based on a trade-off of various factors such as the requirements of the system, the amount of feedback from the receiver to the transmitter, the capabilities of the transmitter and receiver, and so on. The transmitter, receiver, and system are then designed to support and operate in accordance with the selected transmission scheme. This transmission scheme typically has favorable features as well as unfavorable ones, which can impact system performance.

[0005] There is therefore a need in the art for a user terminal capable of achieving improved performance.

SUMMARY

[0006] A user terminal that supports multiple spatial multiplexing (SM) modes for improved performance and greater flexibility is described herein. Spatial multiplexing refers to the transmission of multiple data streams simultaneously via multiple spatial channels of a MIMO channel. The multiple SM modes may include (1) a steered mode that transmits multiple data streams on orthogonal spatial channels and (2) a non-steered mode that transmits multiple data streams from multiple antennas.

[0007] The terminal selects an SM mode to use for data transmission from among the multiple supported SM modes. The SM mode selection may be based on various factors such as the calibration status of the terminal, the amount of data to send, the channel conditions, the capability of the other communicating entity, and so on. For data transmission, multiple data streams are coded and modulated in accordance with their selected rates to obtain multiple data symbol streams. These data symbol streams are then spatially processed in accordance with the selected SM mode to obtain multiple transmit symbol streams. The transmit spatial processing is with a matrix of steering vectors for the steered mode and with an identity matrix for the non-steered mode. The transmit symbol streams are transmitted from multiple antennas and via a first communication link (e.g., uplink).

[0008] For data reception, multiple received symbol streams for a second communication link (e.g., downlink) are spatially processed in accordance with the selected SM mode to obtain multiple recovered data symbol streams. The receive spatial processing may be based on the channel eigenvectors for the steered mode and with a spatial filter matrix for the non-steered mode. The spatial filter matrix may be derived based on various receiver spatial processing techniques, as described below. The recovered data symbol streams are then demodulated and decoded in accordance with their selected rates to obtain multiple decoded data streams for the second link. The terminal also transmits/receives pilots and the selected rates for each link.

[0009] Various aspects, embodiments, and features of the invention are described in further detail below.

BRIEF DESCRIPTION OF THE DRAWINGS

[0010] FIG. 1 shows a MIMO system;

FIG. 2 shows spatial processing at a transmitter and receiver for the steered and non-steered modes;

FIGS. 3 and 4 show spatial processing at an access point and a user terminal for the steered and non-steered modes, respectively;

FIG. 5 shows a block diagram of the access point and user terminal; and

FIG. 6 shows a process for transmitting and receiving data in the MIMO system.

DETAILED DESCRIPTION

[0011] The word “exemplary” is used herein to mean “serving as an example, instance, or illustration.” Any embodiment described herein as “exemplary” is not necessarily to be construed as preferred or advantageous over other embodiments.

[0012] FIG. 1 shows a MIMO system 100 with access points (APs) and user terminals (UTs). For simplicity, only one access point 110 is shown in FIG. 1. An access point is generally a fixed station that communicates with the user terminals and may also be referred to as a base station or some other terminology. A system controller 130 couples to and provides coordination and control for the access points. A user terminal may be fixed or mobile and may also be referred to as a mobile station, a wireless device, or some other terminology. A user terminal may communicate with an access point, in which case the roles of access point and user terminal are established. A user terminal may also communicate peer-to-peer with another user terminal.

[0013] MIMO system 100 may be a time division duplex (TDD) system or a frequency division duplex (FDD) system. For a TDD system, the downlink and uplink share the same frequency band. For an FDD system, the downlink and uplink use different frequency bands. The downlink is the communication link from the access points to the user terminals, and the uplink is the communication link from the user terminals to the access points. MIMO system 100 may also utilize a single carrier or multiple carriers for data transmission.

[0014] Access point 110 and user terminal 120 each support multiple spatial multiplexing (SM) modes for improved performance and greater flexibility. A steered SM mode (or simply, a steered mode) can typically achieve better performance but can only be used if the transmitter has sufficient channel state information (CSI) to

orthogonalize the spatial channels of a MIMO channel via decomposition or some other technique. A non-steered SM mode (or simply, a non-steered mode) requires very little information to simultaneously transmit multiple data streams via the MIMO channel, but performance may not be quite as good as the steered mode. A suitable SM mode may be selected for use based on various factors, as described below.

[0015] Table 1 summarizes some key aspects of the steered and non-steered modes. Each SM mode has different capabilities and requirements.

[0016] For the steered mode, the transmitter transmits a pilot to allow the receiver to estimate the MIMO channel, and the receiver sends back sufficient channel state information to allow the transmitter to derive steering vectors. Either the transmitter or receiver decomposes the MIMO channel into eigenmodes, which may be viewed as orthogonal spatial channels. The receiver also sends back the rate to use for each eigenmode. The transmitter and receiver both perform spatial processing in order to transmit data on the eigenmodes, as described below.

[0017] For the non-steered mode, the transmitter transmits a pilot to allow the receiver to estimate the MIMO channel. The receiver sends back the rate to use for each spatial channel. The transmitter transmits data (e.g., from its antennas) without any spatial processing, and the receiver performs spatial processing to recover the transmitted data. The pilot transmission and spatial processing at the transmitter and receiver for the steered and non-steered modes are described below.

Table 1 – Requirements for Steered and Non-Steered Modes

	Steered Mode	Non-Steered Mode
Pilot	Transmitter transmits a pilot Receiver sends back channel state information used by transmitter to derive steering vectors	Transmitter transmits a pilot
Rate Feedback	Receiver sends back the rate for each eigenmode	Receiver sends back the rate for each spatial channel (e.g., each transmit antenna)
Spatial Processing	Transmitter performs spatial processing with matrix $\underline{\mathbf{V}}$ of steering vectors Receiver performs spatial processing with matrix $\underline{\mathbf{U}}$ of eigenvectors	Transmitter transmits data from each transmit antenna Receiver performs spatial processing with CCMI, MMSE, SIC, and so on (described below)

[0018] In the following description, a user terminal can be the transmitter and/or receiver, and an access point can likewise be the transmitter and/or receiver. Peer-to-peer communications can be supported using the same basic principles.

1. Steered Mode

[0019] A MIMO channel formed by N_T transmit antennas and N_R receive antennas may be characterized by an $N_R \times N_T$ channel response matrix $\underline{\mathbf{H}}$, which may be expressed as:

$$\underline{\mathbf{H}} = \begin{bmatrix} h_{1,1} & h_{1,2} & \Lambda & h_{1,N_T} \\ h_{2,1} & h_{2,2} & \Lambda & h_{2,N_T} \\ \text{M} & \text{M} & \text{O} & \text{M} \\ h_{N_R,1} & h_{N_R,2} & \Lambda & h_{N_R,N_T} \end{bmatrix}, \quad \text{Eq (1)}$$

where entry $h_{i,j}$, for $i=1 \dots N_R$ and $j=1 \dots N_T$, is the coupling (i.e., complex gain) between transmit antenna j and receive antenna i . For simplicity, the MIMO channel is assumed to be full rank with $N_S \leq N_T \leq N_R$.

[0020] Singular value decomposition may be performed on $\underline{\mathbf{H}}$ to obtain N_S eigenmodes of $\underline{\mathbf{H}}$, as follows:

$$\underline{\mathbf{H}} = \underline{\mathbf{U}} \underline{\Sigma} \underline{\mathbf{V}}^H, \quad \text{Eq (2)}$$

where $\underline{\mathbf{U}}$ is an $(N_R \times N_R)$ unitary matrix of left eigenvectors of $\underline{\mathbf{H}}$;

$\underline{\Sigma}$ is an $(N_R \times N_T)$ diagonal matrix of singular values of $\underline{\mathbf{H}}$;

$\underline{\mathbf{V}}$ is an $(N_T \times N_T)$ unitary matrix of right eigenvectors of $\underline{\mathbf{H}}$; and

“ H ” denotes the conjugate transpose.

A unitary matrix $\underline{\mathbf{M}}$ is characterized by the property $\underline{\mathbf{M}}^H \underline{\mathbf{M}} = \underline{\mathbf{I}}$, where $\underline{\mathbf{I}}$ is the identity matrix. The columns of a unitary matrix are orthogonal to one another.

[0021] The right eigenvectors of $\underline{\mathbf{H}}$ are also referred to as steering vectors and may be used for spatial processing by the transmitter to transmit data on the N_S eigenmodes of $\underline{\mathbf{H}}$. The left eigenvectors of $\underline{\mathbf{H}}$ may be used for spatial processing by the receiver to recover the data transmitted on the N_S eigenmodes. The eigenmodes may be viewed as orthogonal spatial channels obtained through decomposition. The diagonal entries of $\underline{\Sigma}$ are the singular values of $\underline{\mathbf{H}}$, which represent the channel gains for the N_S eigenmodes.

[0022] In a practical system, only an estimate of $\underline{\mathbf{H}}$ can be obtained, and only estimates of $\underline{\mathbf{V}}$, $\underline{\Sigma}$ and $\underline{\mathbf{U}}$ can be derived. The N_S spatial channels are also typically not completely orthogonal to one another due to various reasons such as an imperfect channel estimate. For simplicity, the description herein assumes channel estimation and decomposition without errors. Furthermore, the term “eigenmode” covers the case where an attempt is made to orthogonalize the spatial channels using decomposition, even though the attempt may not be fully successful due to, for example, an imperfect channel estimate.

[0023] Table 2 summarizes the spatial processing at the transmitter and the receiver for the steered mode. In Table 2, $\underline{\mathbf{s}}$ is a vector with N_S data symbols to be transmitted on the N_S eigenmodes of $\underline{\mathbf{H}}$, $\underline{\mathbf{x}}_{st}$ is a vector with N_T transmit symbols to be sent from the N_T transmit antennas, $\underline{\mathbf{r}}_{st}$ is a vector with N_R received symbols obtained from the N_R receive antennas, $\hat{\underline{\mathbf{s}}}_{st}$ is a vector with N_S recovered data symbols (i.e., $\hat{\underline{\mathbf{s}}}_{st}$ is an estimate of $\underline{\mathbf{s}}$), and the subscript “ st ” denotes the steered mode. As used herein, a “data symbol” refers to a modulation symbol for data, and a “pilot symbol” refers to a modulation symbol for pilot.

Table 2 – Spatial Processing for Steered Mode

Transmitter	Receiver
$\underline{\mathbf{x}}_{st} = \underline{\mathbf{V}}\underline{\mathbf{s}}$	$\hat{\underline{\mathbf{s}}}_{st} = \underline{\Sigma}^{-1}\underline{\mathbf{U}}^H \underline{\mathbf{r}}_{st}$

[0024] Eigenvalue decomposition may also be performed on a correlation matrix of $\underline{\mathbf{H}}$, which is $\underline{\mathbf{R}} = \underline{\mathbf{H}}^H \underline{\mathbf{H}}$, as follows:

$$\underline{\mathbf{R}} = \underline{\mathbf{H}}^H \underline{\mathbf{H}} = \underline{\mathbf{V}}\underline{\Lambda}\underline{\mathbf{V}}^H, \quad \text{Eq (3)}$$

where $\underline{\Lambda}$ is a diagonal matrix of eigenvalues, which are the squares of the singular values in $\underline{\Sigma}$. The transmitter can perform spatial processing with $\underline{\mathbf{V}}$ to obtain $\underline{\mathbf{x}}_{st}$, and the receiver can perform spatial processing with $\underline{\mathbf{V}}^H \underline{\mathbf{H}}^H$ to obtain $\hat{\underline{\mathbf{s}}}_{st}$.

2. Non-Steered Mode

[0025] For the non-steered mode, the transmitter can transmit one data symbol stream from each transmit antenna. A spatial channel for this mode can correspond to one transmit antenna. The receiver performs spatial processing to separate out and recover the transmitted data symbol streams. The receiver can use various receiver processing techniques such as a channel correlation matrix inversion (CCMI) technique (which is also known as a zero-forcing technique), a minimum mean square error (MMSE) technique, a successive interference cancellation (SIC) technique, and so on

[0026] Table 3 summarizes the spatial processing at the transmitter and the receiver for the non-steered mode. In Table 3, $\underline{\mathbf{x}}_{ns}$ is a vector with N_T data symbols to be sent from the N_T transmit antennas, $\underline{\mathbf{r}}_{ns}$ is a vector with N_R received symbols obtained from the N_R receive antennas, $\underline{\mathbf{M}}_{ccmi}$ is a spatial filter matrix for the CCMI technique, $\underline{\mathbf{M}}_{mmse}$ is a spatial filter matrix for the MMSE technique, $\underline{\mathbf{D}}_{mmse}$ is a diagonal matrix for the MMSE technique (which contains the diagonal elements of $\underline{\mathbf{M}}_{mmse} \underline{\mathbf{H}}$), and the subscript “ns” denotes the non-steered mode.

Table 3 – Spatial Processing for Non-Steered Mode

Transmitter	Receiver	where	
$\underline{\mathbf{x}}_{ns} = \underline{\mathbf{s}}$	$\hat{\underline{\mathbf{s}}}_{ccmi} = \underline{\mathbf{M}}_{ccmi} \underline{\mathbf{r}}_{ns}$	$\underline{\mathbf{M}}_{ccmi} = [\underline{\mathbf{H}}^H \underline{\mathbf{H}}]^{-1} \underline{\mathbf{H}}^H$	CCMI
	$\hat{\underline{\mathbf{s}}}_{mmse} = \underline{\mathbf{D}}_{mmse}^{-1} \underline{\mathbf{M}}_{mmse} \underline{\mathbf{r}}_{ns}$	$\underline{\mathbf{M}}_{mmse} = \underline{\mathbf{H}}^H [\underline{\mathbf{H}} \underline{\mathbf{H}}^H + \sigma^2 \underline{\mathbf{I}}]^{-1}$ and $\underline{\mathbf{D}}_{mmse} = \text{diag} [\underline{\mathbf{M}}_{mmse} \underline{\mathbf{H}}]$	MMSE
	$\hat{\underline{\mathbf{s}}}_{sic}^\lambda = \underline{\mathbf{M}}_{sic}^\lambda \underline{\mathbf{r}}_{sic}^\lambda$	$\underline{\mathbf{M}}_{sic}^\lambda = \underline{\mathbf{M}}_{ccmi}^\lambda$ or $(\underline{\mathbf{D}}_{mmse}^\lambda)^{-1} \underline{\mathbf{M}}_{mmse}^\lambda$	SIC

For simplicity, the MIMO channel noise $\underline{\mathbf{n}}$ is assumed to be additive white Gaussian noise (AWGN) with zero mean, a variance of σ^2 , and an autocovariance matrix of $\underline{\varphi}_{nn} = E[\underline{\mathbf{n}}\underline{\mathbf{n}}^H] = \sigma^2 \underline{\mathbf{I}}$.

[0027] For the SIC technique, the receiver processes the N_R received symbol streams in N_S successive stages to recover one data symbol stream in each stage. For each stage λ , where $\lambda = 1 \dots N_S$, the receiver initially performs spatial processing on N_R input symbol streams for stage λ using the CCMI, MMSE, or some other technique and obtains one recovered data symbol stream. The N_R received symbol streams are the N_R input symbol streams for stage 1. The receiver further processes (e.g., demodulates, deinterleaves, and decodes) the recovered data symbol stream for stage λ to obtain a decoded data stream, estimates the interference this stream causes to the other data symbol streams not yet recovered, and cancels the estimated interference from the N_R input symbol streams for stage λ to obtain N_R input symbol streams for stage $\lambda+1$. The receiver then repeats the same processing on the N_R input symbol streams for stage $\lambda+1$ to recover another data symbol stream.

[0028] For each stage λ , the SIC receiver derives a spatial filter matrix $\underline{\mathbf{M}}_{sic}^\lambda$ for that stage based on a reduced channel response matrix $\underline{\mathbf{H}}^\lambda$ and using the CCMI, MMSE, or some other technique. The reduced matrix $\underline{\mathbf{H}}^\lambda$ is obtained by removing $\lambda-1$ columns in the original matrix $\underline{\mathbf{H}}$ corresponding to the $\lambda-1$ data symbol streams already recovered. The matrix $\underline{\mathbf{M}}_{sic}^\lambda$ has dimensionality of $(N_T - \lambda + 1) \times N_R$. Since $\underline{\mathbf{H}}^\lambda$ is different for each stage, $\underline{\mathbf{M}}_{sic}^\lambda$ is also different for each stage.

[0029] The receiver may also use other receiver spatial processing techniques to recover the transmitted data symbol streams.

[0030] FIG. 2 shows the spatial processing at the transmitter and receiver for the steered and non-steered modes. At the transmitter, the data vector \underline{s} is multiplied with either the matrix \underline{V} for the steered mode or the identity matrix \underline{I} for the non-steered mode by a unit 220 to obtain the transmit symbol vector \underline{x} . At the receiver, the received symbol vector \underline{r} is multiplied with either the matrix \underline{U}^H for the steered mode or the spatial filter matrix \underline{M} for the non-steered mode by a unit 260 to obtain a detected symbol vector $\underline{\tilde{s}}$, which is an unnormalized estimate of \underline{s} . The matrix \underline{M} may be derived based on the CCMI, MMSE, or some other technique. The vector $\underline{\tilde{s}}$ is further scaled with either the diagonal matrix $\underline{\Sigma}^{-1}$ for the steered mode or a diagonal matrix \underline{D}^{-1} for the non-steered mode to obtain the recovered data symbol vector $\underline{\hat{s}}$, where $\underline{D}^{-1} = \underline{I}$ for the CCMI technique and $\underline{D}^{-1} = \underline{D}_{mmse}^{-1}$ for the MMSE technique.

3. Overhead for Steered and Non-Steered Nodes

[0031] The steered and non-steered modes have different pilot and overhead requirements, as shown in

[0032] Table 1 and described below.

A. Pilot Transmission

[0033] For both the steered and non-steered modes, the transmitter can transmit a MIMO pilot (which is an unsteered pilot) to allow the receiver to estimate the MIMO channel and obtain the matrix \underline{H} . The MIMO pilot comprises N_T orthogonal pilot transmissions sent from N_T transmit antennas, where orthogonality may be achieved in time, frequency, code, or a combination thereof. For code orthogonality, the N_T pilot transmissions can be sent simultaneously from the N_T transmit antennas, with the pilot transmission from each antenna being “covered” with a different orthogonal (e.g., Walsh) sequence. The receiver “discovers” the received pilot symbols for each receive antenna i with the same N_T orthogonal sequences used by the transmitter to obtain estimates of the complex channel gain between receive antenna i and each of the N_T transmit antennas. The covering at the transmitter and the discovering at the receiver are performed in similar manner as for a Code Division Multiple Access (CDMA) system. For frequency orthogonality, the N_T pilot transmissions for the N_T transmit antennas can be sent simultaneously on different subbands of the overall system bandwidth. For time

orthogonality, the N_T pilot transmissions for the N_T transmit antennas can be sent in different time slots. In any case, the orthogonality among the N_T pilot transmissions allows the receiver to distinguish the pilot transmission from each transmit antenna.

[0034] For the steered mode, the receiver sends back sufficient channel state information to allow the transmitter to derive the steering vectors. The receiver may send this information in a direct form (e.g., by sending the entries of $\underline{\mathbf{V}}$) or in an indirect form (e.g., by transmitting a steered or unsteered pilot).

B. Rate Selection/Control

[0035] The receiver can estimate the received signal-to-noise-and-interference ratio (SNR) for each spatial channel, which can correspond to an eigenmode for the steered mode or a transmit antenna for the non-steered mode. The received SNR is dependent on the SM mode and spatial processing technique used by the transmitter and receiver.

[0036] Table 4 summarizes the received SNR for the steered and non-steered modes. In Table 4, P_m is the transmit power used for spatial channel m , σ^2 is the noise variance, σ_m is the singular value for eigenmode m (i.e., the m -th diagonal element of $\underline{\Sigma}$), r_{mm} is the m -th diagonal element of $\underline{\mathbf{R}}$ (which is $\underline{\mathbf{R}} = \underline{\mathbf{H}}^H \underline{\mathbf{H}}$), q_{mm} is the m -th diagonal element of $\underline{\mathbf{Q}}$, and γ_m is the SNR for spatial channel m . The received SNRs for the SIC technique are dependent on the spatial processing technique (e.g., CCMI or MMSE) and the order in which the data streams are recovered. An operating SNR can be defined as being equal to the received SNR plus an SNR back-off factor. The SNR back-off factor can be set to a positive value to account for estimation error, SNR fluctuation over time, and so on, but may also be set to zero.

Table 4 – Received SNR

Steered Mode	Non-Steered Mode	
	CCMI	MMSE
$\gamma_{st,m} = \frac{P_m \cdot \sigma_m^2}{\sigma^2}$	$\gamma_{ccmi,m} = \frac{P_m}{r_{mm} \cdot \sigma^2}$	$\gamma_{mmse,m} = \frac{q_{mm}}{1 - q_{mm}} P_m$

[0037] The MIMO system may support a set of rates. Each non-zero rate is associated with a particular data rate or spectral efficiency, a particular coding scheme, a particular modulation scheme, and a particular SNR required to achieve a target level of performance (e.g., one percent packet error rate (PER)). The required SNR for each rate may be determined by computer simulation, empirical measurement, and so on, and with an assumption of an AWGN channel. A look-up table (LUT) can store the rates supported by the system and their required SNRs. For each spatial channel, the highest rate in the look-up table with a required SNR that is equal to or less than the operating SNR of the spatial channel is selected as the rate to use for the spatial channel.

[0038] Closed-loop rate control may be used for each spatial channel or a combination of spatial channels. The receiver can estimate the received SNR for each spatial channel, select the proper rate for the spatial channel, and send back the selected rate. The transmitter can transmit each data symbol stream at the selected rate.

C. Mode Selection

[0039] User terminal 120 can use either the steered or non-steered mode at any given moment for communication. The mode selection may be made based on various factors such as the following.

[0040] Overhead – The steered mode requires more overhead than the non-steered mode. For the steered mode, the receiver needs to send back sufficient channel state information as well as the rates for the N_S eigenmodes. In some instances, the additional CSI overhead cannot be supported or is not justified. For the non-steered mode, the receiver only needs to send back the rates for the spatial channels, which is much less overhead.

[0041] Amount of Data – The steered mode is generally more efficient but also requires more setup steps (e.g., channel estimation, singular value decomposition, and CSI

feedback). If only a small amount of data needs to be sent, then it may be quicker and more efficient to transmit this data using the non-steered mode.

[0042] Capability – A user terminal may communicate peer-to-peer with another user terminal that supports only one mode (e.g., either the steered or non-steered mode). In this case, the two terminals can communicate using a common mode supported by both user terminals.

[0043] Channel Conditions – The steered mode may be more easily supported for static channels, slow varying channels, and channels with a strong line-of-site component (e.g., a Rician channel).

[0044] Receiver SNR – The steered mode provides better performance in low SNR conditions. A user terminal may elect to use steered mode when the SNR drops below some threshold.

[0045] Calibration Status – The steered mode may be selected for use if the transmitter and receiver are “calibrated” such that the downlink and uplink channel responses are reciprocal of one another. Reciprocal downlink and uplink can simplify the pilot transmission and spatial processing for both the transmitter and receiver for the steered mode, as described below.

[0046] A user terminal that is not mobile and is communicating with the same access point may use the steered mode much of the time. A user terminal that is mobile and communicating with different entities (e.g., different access points and/or other user terminals) may use the non-steered mode, until such time that it is more advantageous to use the steered mode. A user terminal may also switch between the steered and non-steered modes, as appropriate. For example, a user terminal may use the non-steered mode for small data bursts (or short data sessions) and at the start of long data bursts (or long data sessions), and may use the steered mode for the remaining portion of the long data bursts. As another example, a user terminal may use the steered mode for relatively static channel conditions and may use the non-steered mode when the channel conditions change more rapidly.

4. TDD MIMO System

[0047] A multi-mode user terminal for an exemplary MIMO wireless local area network (WLAN) system is described below. The MIMO WLAN system utilizes orthogonal frequency division multiplexing (OFDM), which is a multi-carrier modulation technique

that effectively partitions the overall system bandwidth into multiple (N_F) orthogonal subbands. With OFDM, each subband is associated with a respective carrier that may be modulated with data.

[0048] The exemplary MIMO WLAN system is a TDD system. A high degree of correlation normally exists between the downlink and uplink channel responses for the TDD system since these links share the same frequency band. However, the responses of the transmit/receive chains at the access point are typically not the same as the responses of the transmit/receive chains at the user terminal. The differences can be determined and accounted for via calibration. The overall downlink and uplink channel responses may then be assumed to be reciprocal (i.e., transpose) of each other. The channel estimation and spatial processing for the steered mode can be simplified with reciprocal downlink and uplink.

[0049] FIG. 3 shows the transmit/receive chains at access point 110 and user terminal 120. At access point 110, transmit chain 324 and receive chain 334 are modeled by matrices $\underline{\mathbf{T}}_{\text{ap}}(k)$ and $\underline{\mathbf{R}}_{\text{ap}}(k)$, respectively, for each subband k . At user terminal 120, transmit chain 364 and receive chain 354 are modeled by matrices $\underline{\mathbf{T}}_{\text{ut}}(k)$ and $\underline{\mathbf{R}}_{\text{ut}}(k)$, respectively, for each subband k .

[0050] Table 5 summarizes the calibration and singular value decomposition for the downlink and uplink in the TDD MIMO WLAN system. The “effective” downlink and uplink channel responses, $\underline{\mathbf{H}}_{\text{cdn}}(k)$ and $\underline{\mathbf{H}}_{\text{cup}}(k)$, include the responses of the appropriate transmit and receive chains. Diagonal correction matrices $\underline{\mathbf{K}}_{\text{ap}}(k)$ and $\underline{\mathbf{K}}_{\text{ut}}(k)$ are obtained by performing calibration with MIMO pilots transmitted by both the access point and user terminal. The “calibrated” downlink and uplink channel responses, $\underline{\mathbf{H}}_{\text{cdn}}(k)$ and $\underline{\mathbf{H}}_{\text{cup}}(k)$, include the correction matrices and are reciprocal of one another (i.e., $\underline{\mathbf{H}}_{\text{cup}}(k) = \underline{\mathbf{H}}_{\text{cdn}}^T(k)$, where “ T ” denotes the transpose).

Table 5 – Channel Responses for TDD MIMO WLAN System

	Downlink	Uplink
Effective Channel Response	$\underline{\mathbf{H}}_{\text{edn}}(k) = \underline{\mathbf{R}}_{\text{ut}}(k) \underline{\mathbf{H}}(k) \underline{\mathbf{T}}_{\text{ap}}(k)$	$\underline{\mathbf{H}}_{\text{cup}}(k) = \underline{\mathbf{R}}_{\text{ap}}(k) \underline{\mathbf{H}}^T(k) \underline{\mathbf{T}}_{\text{ut}}(k)$
Correction Matrix	$\underline{\mathbf{K}}_{\text{ap}}(k) = \underline{\mathbf{T}}_{\text{ap}}^{-1}(k) \underline{\mathbf{R}}_{\text{ap}}(k)$	$\underline{\mathbf{K}}_{\text{ut}}(k) = \underline{\mathbf{T}}_{\text{ut}}^{-1}(k) \underline{\mathbf{R}}_{\text{ut}}(k)$
Calibrated Channel Response	$\underline{\mathbf{H}}_{\text{cdn}}(k) = \underline{\mathbf{H}}_{\text{edn}}(k) \underline{\mathbf{K}}_{\text{ap}}(k)$	$\underline{\mathbf{H}}_{\text{cup}}(k) = \underline{\mathbf{H}}_{\text{cup}}(k) \underline{\mathbf{K}}_{\text{ut}}(k)$
Singular Value Decomposition	$\underline{\mathbf{H}}_{\text{cdn}}(k) = \underline{\mathbf{V}}_{\text{ut}}^*(k) \underline{\Sigma}(k) \underline{\mathbf{U}}_{\text{ap}}^T(k)$	$\underline{\mathbf{H}}_{\text{cup}}(k) = \underline{\mathbf{U}}_{\text{ap}}(k) \underline{\Sigma}(k) \underline{\mathbf{V}}_{\text{ut}}^H(k)$

[0051] Because $\underline{\mathbf{H}}_{\text{cup}}(k)$ and $\underline{\mathbf{H}}_{\text{cdn}}(k)$ are reciprocal, the matrices $\underline{\mathbf{V}}_{\text{ut}}^*(k)$ and $\underline{\mathbf{U}}_{\text{ap}}^*(k)$ of left and right eigenvectors of $\underline{\mathbf{H}}_{\text{cdn}}(k)$ are the complex conjugate of the matrices $\underline{\mathbf{V}}_{\text{ut}}(k)$ and $\underline{\mathbf{U}}_{\text{ap}}(k)$ of right and left eigenvectors of $\underline{\mathbf{H}}_{\text{cup}}(k)$. The matrix $\underline{\mathbf{U}}_{\text{ap}}(k)$ can be used by access point 110 for both transmit and receive spatial processing. The matrix $\underline{\mathbf{V}}_{\text{ut}}(k)$ can be used by user terminal 120 for both transmit and receive spatial processing.

[0052] Singular value decomposition may be performed independently for each of the N_F subbands. For each subband, the singular values in $\underline{\Sigma}(k)$ may be ordered from largest to smallest, and the eigenvectors in $\underline{\mathbf{V}}(k)$ and $\underline{\mathbf{U}}(k)$ may be ordered correspondingly. A “wideband” eigenmode may be defined as the set of same-order eigenmodes for all N_F subbands after the ordering. The decomposition only needs to be performed by either user terminal 120 or access point 110. If performed by user terminal 120, then the matrices $\underline{\mathbf{U}}_{\text{ap}}(k)$, for $k = 1 \dots N_F$, may be provided to access point 110 in either a direct form (e.g., by sending entries of $\underline{\mathbf{U}}_{\text{ap}}(k)$) or an indirect form (e.g., by transmitting a steered pilot).

[0053] Table 6 summarizes the spatial processing at access point 110 and user terminal 120 for data transmission and reception on the downlink and uplink in the TDD MIMO WLAN system for the steered mode. In Table 6, the subscript “up” denotes the uplink, and the subscript “dn” denotes the downlink.

Table 6 – Spatial Processing for Steered Mode in TDD MIMO WLAN System

	Downlink	Uplink
Access Point	Transmit: $\underline{\mathbf{x}}_{\text{dn}}(k) = \underline{\mathbf{K}}_{\text{ap}}(k) \underline{\mathbf{U}}_{\text{ap}}^*(k) \underline{\mathbf{s}}_{\text{dn}}(k)$	Receive: $\hat{\underline{\mathbf{s}}}_{\text{up}}(k) = \underline{\Sigma}^{-1}(k) \underline{\mathbf{U}}_{\text{ap}}^H(k) \underline{\mathbf{r}}_{\text{up}}(k)$
User Terminal	Receive: $\hat{\underline{\mathbf{s}}}_{\text{dn}}(k) = \underline{\Sigma}^{-1}(k) \underline{\mathbf{V}}_{\text{ut}}^T(k) \underline{\mathbf{r}}_{\text{dn}}(k)$	Transmit: $\underline{\mathbf{x}}_{\text{up}}(k) = \underline{\mathbf{K}}_{\text{ut}}(k) \underline{\mathbf{V}}_{\text{ut}}(k) \underline{\mathbf{s}}_{\text{up}}(k)$

[0054] For the steered mode, the access point can transmit a MIMO pilot on the downlink. The user terminal can estimate the calibrated downlink channel based on the MIMO pilot, perform singular value decomposition, and transmit a steered pilot on the uplink using the matrix $\underline{\mathbf{V}}_{\text{ut}}(k)$. A steered pilot is a pilot transmitted on the eigenmodes using the same steering vectors that are used for data transmission on the eigenmodes. The access point can directly estimate the matrix $\underline{\mathbf{U}}_{\text{ap}}(k)$ based on the uplink steered pilot. Pilots may also be transmitted in other manners for the steered mode. For example, the user terminal can transmit the MIMO pilot, and the access point can transmit the steered pilot. As another example, the access point and user terminal can both transmit MIMO pilots.

[0055] For the non-steered mode, the transmitter (either the access point or user terminal) can transmit a MIMO pilot along with the data transmission. The receiver performs spatial processing (e.g., with CCMI, MMSE, SIC, or some other technique) to recover the data symbol streams, as described above.

[0056] Table 7 summarizes an embodiment of the pilot transmission and spatial processing for the steered and non-steered modes for the TDD MIMO WLAN system.

Table 7 – Data Transmission in TDD MIMO WLAN System

	Steered Mode	Non-Steered Mode
Calibration	Calibration is performed	Calibration is not required
Downlink Data Transmission	AP transmits a MIMO pilot UT transmits a steered pilot	AP transmits a MIMO pilot
	UT sends the rate for each downlink wideband eigenmode	UT sends the rate for each downlink wideband spatial channel
	AP transmits data with $\underline{\mathbf{U}}_{\text{ap}}(k)$ UT receives data with $\underline{\mathbf{V}}_{\text{ut}}(k)$	AP transmits data from each antenna UT receives data with CCMI, MMSE, SIC, and so on
Uplink Data Transmission	AP transmits a MIMO pilot UT transmits a steered pilot	UT transmits a MIMO pilot
	AP sends the rate for each uplink wideband eigenmode	AP sends the rate for each uplink wideband spatial channel
	UT transmits data with $\underline{\mathbf{V}}_{\text{ut}}(k)$ AP receives data with $\underline{\mathbf{U}}_{\text{ap}}(k)$	UT transmits data from each antenna AP receives data with CCMI, MMSE, SIC, and so on

[0057] For both the steered and non-steered modes, the receiver (either the access point or user terminal) can estimate the average received SNR for each wideband spatial channel, for example, by averaging the received SNRs (in dB) for the N_F subbands of the wideband spatial channel. A wideband spatial channel can correspond to a wideband eigenmode for the steered mode or a transmit antenna for the non-steered mode. The receiver then computes an operating SNR for each wideband spatial channel as the sum of the average received SNR plus the SNR back-off factor. The receiver then selects the rate for each wideband spatial channel based on the operating SNR and the look-up table of supported rates and their required SNRs.

[0058] FIG. 3 shows the spatial processing at access point 110 and user terminal 120 for downlink and uplink data transmission for the steered mode in the MIMO WLAN system. For the downlink, at access point 110, the data symbol vector $\underline{\mathbf{s}}_{\text{dn}}(k)$ is multiplied with the matrix $\underline{\mathbf{U}}_{\text{ap}}^*(k)$ by a unit 320 and further scaled with the correction matrix $\underline{\mathbf{K}}_{\text{ap}}(k)$ by a unit 322 to obtain the transmit symbol vector $\underline{\mathbf{x}}_{\text{dn}}(k)$ for the

downlink. At user terminal 120, the received symbol vector $\mathbf{r}_{dn}(k)$ is multiplied with the matrix $\mathbf{V}_{ut}^T(k)$ by a unit 360 and further scaled with the matrix $\mathbf{\Sigma}^{-1}(k)$ by a unit 362 to obtain the recovered data symbol vector $\hat{\mathbf{s}}_{dn}(k)$ for the downlink.

[0059] For the uplink, at user terminal 120, the data symbol vector $\mathbf{s}_{up}(k)$ is multiplied with the matrix $\mathbf{V}_{ut}(k)$ by a unit 390 and further scaled with the correction matrix $\mathbf{K}_{ut}(k)$ by a unit 392 to obtain the transmit symbol vector $\mathbf{x}_{up}(k)$ for the uplink. At access point 110, the received symbol vector $\mathbf{r}_{up}(k)$ is multiplied with the matrix $\mathbf{U}_{ap}^H(k)$ by a unit 340 and further scaled with the matrix $\mathbf{\Sigma}^{-1}(k)$ by a unit 342 to obtain the recovered data symbol vector $\hat{\mathbf{s}}_{up}(k)$ for the uplink.

[0060] FIG. 4 shows the spatial processing at access point 110 and user terminal 120 for downlink and uplink data transmission for the non-steered mode in the MIMO WLAN system. For the downlink, at access point 110, the data symbol vector $\mathbf{s}_{dn}(k)$ is multiplied with the identity matrix \mathbf{I} by a unit 420 to obtain the transmit symbol vector $\mathbf{x}_{dn}(k)$ for the downlink. At user terminal 120, the received symbol vector $\mathbf{r}_{dn}(k)$ is multiplied with a spatial filter matrix $\mathbf{M}_{ut}(k)$ by a unit 460 and further scaled with a diagonal matrix $\mathbf{D}_{ut}^{-1}(k)$ by a unit 462 to obtain the recovered data symbol vector $\hat{\mathbf{s}}_{dn}(k)$ for the downlink. The matrices $\mathbf{M}_{ut}(k)$ and $\mathbf{D}_{ut}^{-1}(k)$ are derived based on the effective downlink channel response matrix $\mathbf{H}_{edn}(k)$ and using the CCMI, MMSE, SIC, or some other technique.

[0061] For the uplink, at user terminal 120, the data symbol vector $\mathbf{s}_{up}(k)$ is multiplied with the identity matrix \mathbf{I} by a unit 490 to obtain the transmit symbol vector $\mathbf{x}_{up}(k)$ for the uplink. At access point 110, the received symbol vector $\mathbf{r}_{up}(k)$ is multiplied with a spatial filter matrix $\mathbf{M}_{ap}(k)$ by a unit 440 and further scaled with a diagonal matrix $\mathbf{D}_{ap}^{-1}(k)$ by a unit 442 to obtain the recovered data symbol vector $\hat{\mathbf{s}}_{up}(k)$ for the uplink. The matrices $\mathbf{M}_{ap}(k)$ and $\mathbf{D}_{ap}^{-1}(k)$ are derived based on the effective uplink channel response matrix $\mathbf{H}_{eup}(k)$ and using the CCMI, MMSE, SIC, or some other technique.

[0062] FIG. 5 shows a block diagram of access point 110 and user terminal 120. On the downlink, at access point 110, a transmit (TX) data processor 514 receives traffic

data from a data source 512 and control data from a controller 530. TX data processor 514 processes (e.g., encodes, interleaves, and symbol maps) each of N_S data streams based on the coding and modulation schemes corresponding to the rate selected for the stream to obtain a data symbol stream. A TX spatial processor 520 receives N_S data symbol streams from TX data processor 514, performs spatial processing (as required) on the data symbols, multiplexes in pilot symbols, and provides N_{ap} transmit symbol streams for the N_{ap} antennas. The processing by TX spatial processor 520 is dependent on whether the steered or non-steered mode is selected for use and may be performed as described above. Each transmitter unit (TMTR) 522 receives and processes (e.g., OFDM modulates and conditions) a respective transmit symbol stream to generate a downlink signal. N_{ap} transmitter units 522a through 522ap provide N_{ap} downlink signals for transmission from N_{ap} antennas 524a through 524ap, respectively.

[0063] At user terminal 120, N_{ut} antennas 552a through 552ut receive the N_{ap} downlink signals, and each antenna provides a received signal to a respective receiver unit (RCVR) 554. Each receiver unit 554 performs processing (e.g., conditioning and OFDM demodulation) complementary to that performed by transmitter units 522 and provides a stream of received symbols. A receive (RX) spatial processor 560 performs spatial processing on N_{ut} received symbol streams from N_{ut} receiver units 554 and provides N_S streams of recovered data symbols. The processing by RX spatial processor 560 is dependent on whether the steered or non-steered mode is selected for use and may be performed as described above. An RX data processor 570 processes (e.g., demaps, deinterleaves, and decodes) the N_S recovered data symbol streams to obtain N_S decoded data streams, which may be provided to a data sink 572 for storage and/or a controller 580 for further processing.

[0064] A channel estimator 578 estimates the downlink channel response based on received pilot symbols and provides channel estimates, which may include channel gain estimates, SNR estimates, and so on. Controller 580 receives the channel estimates, derives the matrices used by RX spatial processor 560 and a TX spatial processor 590 for spatial processing, and determines a suitable rate for each data symbol stream sent on the downlink. The rates and uplink data are processed by a TX data processor 588, spatially processed (as required) by TX spatial processor 590, multiplexed with pilot symbols, conditioned by N_{ut} transmitter units 554a through 554ut, and transmitted via antennas 552a through 552ut.

- [0065] At access point 110, the N_{ut} transmitted uplink signals are received by antennas 524, conditioned and demodulated by receiver units 522, and processed by an RX spatial processor 540 and an RX data processor 542. The rates are provided to controller 530 and used to control data transmission on the downlink.
- [0066] Access point 110 and user terminal 120 may perform similar or different processing for uplink data and pilot transmission.
- [0067] Controllers 530 and 580 control the operation of various processing units at access point 110 and user terminal 120, respectively. SM mode selectors 534 and 584 select the appropriate spatial multiplexing mode to use for access point 110 and user terminal 120, respectively, based on various factors such as those described above. Memory units 532 and 582 store data and program codes used by controllers 530 and 580, respectively.
- [0068] FIG. 6 shows a flow diagram of a process 600 for transmitting and receiving data in the MIMO system. Process 600 may be performed by a user terminal and an access point for data transmission on the downlink and uplink.
- [0069] Initially, an SM mode is selected from among multiple supported SM modes, which may include the steered and non-steered modes described above (step 612). The mode selection may be based on the calibration status of the terminal, the amount of data to send, the SNR and/or channel conditions, the capability of the other communicating entity, and so on. The selected SM mode may also change during a data session.
- [0070] For data transmission (block 620), multiple data streams for a first communication link (e.g., the uplink) are coded and modulated in accordance with their selected rates to obtain multiple data symbol streams for the first link (step 622). These data symbol streams are then spatially processed in accordance with the selected SM mode to obtain multiple transmit symbol streams for transmission from multiple antennas and via the first link (step 624). The transmit spatial processing is with a matrix of steering vectors for the steered mode and with the identity matrix for the non-steered mode.
- [0071] For data reception (block 630), multiple received symbol streams, obtained from the multiple antennas for a second communication link (e.g., the downlink), are spatially processed in accordance with the selected SM mode to obtain multiple recovered data symbol streams (step 632). The receive spatial processing is with a matrix of

eigenvectors for the steered mode and a spatial filter matrix for the non-steered mode. The spatial filter matrix may be derived based on the CCMI, MMSE, SIC, or some other technique. The recovered data symbol streams are then demodulated and decoded in accordance with their selected rates to obtain multiple decoded data streams for the second link (step 634).

[0072] The data transmission in block 620 and the data reception in block 630 may occur simultaneously or at different times. Pilots and rates are also transmitted and received in order to support data transmission and reception with the selected SM mode.

[0073] The multi-mode terminal and access point and the data transmission/reception techniques described herein may be implemented by various means. For example, these entities and techniques may be implemented in hardware, software, or a combination thereof. For a hardware implementation, the processing units for these entities and techniques may be implemented within one or more application specific integrated circuits (ASICs), digital signal processors (DSPs), digital signal processing devices (DSPDs), programmable logic devices (PLDs), field programmable gate arrays (FPGAs), processors, controllers, micro-controllers, microprocessors, other electronic units designed to perform the functions described herein, or a combination thereof.

[0074] For a software implementation, the techniques described herein may be implemented with modules (e.g., procedures, functions, and so on) that perform the functions described herein. The software codes may be stored in a memory unit (e.g., memory units 532 and 582 in FIG. 5) and executed by a processor (e.g., controllers 530 and 580). The memory unit may be implemented within the processor or external to the processor, in which case it can be communicatively coupled to the processor via various means as is known in the art.

[0075] Headings are included herein for reference and to aid in locating certain sections. These headings are not intended to limit the scope of the concepts described therein under, and these concepts may have applicability in other sections throughout the entire specification.

[0076] The previous description of the disclosed embodiments is provided to enable any person skilled in the art to make or use the present invention. Various modifications to these embodiments will be readily apparent to those skilled in the art, and the generic principles defined herein may be applied to other embodiments without departing from the spirit or scope of the invention. Thus, the present invention is not intended to be

limited to the embodiments shown herein but is to be accorded the widest scope consistent with the principles and novel features disclosed herein.

[0077] **WHAT IS CLAIMED IS:**

CLAIMS

1. A terminal in a wireless multiple-input multiple-output (MIMO) communication system, comprising:

a mode selector operable to select a spatial multiplexing mode from among a plurality of spatial multiplexing modes supported by the terminal, wherein each of the plurality of spatial multiplexing modes supports simultaneous transmission of multiple data symbol streams via multiple spatial channels of a MIMO channel formed with a plurality of antennas at the terminal;

a transmit spatial processor operable to spatially process a first plurality of data symbol streams in accordance with the selected spatial multiplexing mode to obtain a plurality of transmit symbol streams for transmission from the plurality of antennas and via a first communication link; and

a receive spatial processor operable to spatially process a plurality of received symbol streams, obtained from the plurality of antennas, in accordance with the selected spatial multiplexing mode to obtain a plurality of recovered data symbol streams, which are estimates of a second plurality of data symbol streams sent via a second communication link.

2. The terminal of claim 1, wherein the plurality of spatial multiplexing modes include a steered mode and a non-steered mode.

3. The terminal of claim 2, wherein the steered mode supports simultaneous transmission of multiple data symbol streams via multiple orthogonal spatial channels of the MIMO channel, and wherein the non-steered mode supports simultaneous transmission of multiple data symbol streams from the plurality of antennas.

4. The terminal of claim 2, wherein

the transmit spatial processor is operable to multiply the first plurality of data symbol streams with a matrix of steering vectors for the steered mode and with an identity matrix for the non-steered mode, and

the receive spatial processor is operable to multiply the plurality of received symbol streams with a matrix of eigenvectors for the steered mode and with a spatial filter matrix for the non-steered mode.

5. The terminal of claim 4, further comprising:

a channel estimator operable to estimate a channel response of the second communication link; and

a controller operable to derive the spatial filter matrix based on the estimated channel response for the second communication link.

6. The terminal of claim 5, wherein the controller is operable to derive the spatial filter matrix based on a channel correlation matrix inversion (CCMI) technique or a minimum mean square error (MMSE) technique.

7. The terminal of claim 5, wherein the controller is operable to derive the spatial filter matrix based on a successive interference cancellation (SIC) technique and using a channel correlation matrix inversion (CCMI) technique or a minimum mean square error (MMSE) technique.

8. The terminal of claim 2, further comprising:

a transmit data processor operable to code and modulate a first plurality of data streams in accordance with a first plurality of rates to obtain the first plurality of data symbol streams for the first communication link; and

a receive data processor operable to demodulate and decode the plurality of recovered data symbol streams in accordance with a second plurality of rates to obtain a plurality of decoded data streams for the second communication link.

9. The terminal of claim 8, wherein the first plurality of rates are for a plurality of eigenmodes of the MIMO channel for the steered mode and are for the plurality of antennas for the non-steered mode.

10. The terminal of claim 2, wherein the mode selector is operable to select the steered mode if the terminal is calibrated and the non-steered mode if the terminal is not calibrated, and wherein channel response of the second communication link is reciprocal of channel response of the first communication link if the terminal is calibrated.

11. The terminal of claim 2, wherein the mode selector is operable to select the steered mode or the non-steered mode based on an amount of data to send, channel conditions, capability of an entity in communication with the terminal, or a combination thereof.

12. The terminal of claim 2, wherein the mode selector is operable to select the non-steered mode for a first portion of a data session and to select the steered mode for a remaining portion of the data session.

13. The terminal of claim 2, wherein the mode selector is operable to select the steered mode or the non-steered mode based on received signal-to-noise-and-interference ratio (SNR).

14. The terminal of claim 2, wherein the transmit spatial processor is further operable to multiplex a steered pilot for the steered mode and an unsteered pilot for the non-steered mode, wherein the steered pilot is transmitted on eigenmodes of the MIMO channel, and wherein the unsteered pilot comprises a plurality of orthogonal pilot transmissions from the plurality of antennas.

15. The terminal of claim 2, wherein the transmit spatial processor is further operable to multiplex an unsteered pilot for both the steered and non-steered modes, and wherein the unsteered pilot comprises a plurality of orthogonal pilot transmissions from the plurality of antennas.

16. The terminal of claim 1 and operable to communicate with an access point in the MIMO system.

17. The terminal of claim 1 and operable to communicate peer-to-peer with another terminal in the MIMO system.

18. The terminal of claim 1, wherein the MIMO system utilizes orthogonal frequency division multiplexing (OFDM), and wherein the transmit and receive spatial processors are operable to perform spatial processing for each of a plurality of subbands.

19. The terminal of claim 1, wherein the MIMO system is a time division duplex (TDD) system.

20. A method of processing data in a wireless multiple-input multiple-output (MIMO) communication system, comprising:

selecting a spatial multiplexing mode from among a plurality of spatial multiplexing modes, wherein each of the plurality of spatial multiplexing modes supports simultaneous transmission of multiple data symbol streams via multiple spatial channels of a MIMO channel;

spatially processing a first plurality of data symbol streams in accordance with the selected spatial multiplexing mode to obtain a plurality of transmit symbol streams for transmission from a plurality of antennas and via a first communication link; and

spatially processing a plurality of received symbol streams, obtained from the plurality of antennas, in accordance with the selected spatial multiplexing mode to obtain a plurality of recovered data symbol streams, which are estimates of a second plurality of data symbol streams sent via a second communication link.

21. The method of claim 20, wherein the plurality of spatial multiplexing modes include a steered mode and a non-steered mode, the steered mode supporting simultaneous transmission of multiple data symbol streams via multiple orthogonal spatial channels of the MIMO channel, and the non-steered mode supporting simultaneous transmission of multiple data symbol streams from the plurality of antennas.

22. The method of claim 21, wherein the first plurality of data symbol streams are multiplied with a matrix of steering vectors for the steered mode and with an identity matrix for the non-steered mode, and wherein the plurality of received symbol streams are multiplied with a matrix of eigenvectors for the steered mode and with a spatial filter matrix for the non-steered mode.

23. The method of claim 22, further comprising:
estimating a channel response of the second communication link; and
deriving the spatial filter matrix based on the estimated channel response for the second communication link.

24. The method of claim 23, wherein the spatial filter matrix is derived based on a channel correlation matrix inversion (CCMI) technique, a minimum mean square error (MMSE) technique, or a successive interference cancellation (SIC) technique.

25. An apparatus in a wireless multiple-input multiple-output (MIMO) communication system, comprising:

means for selecting a spatial multiplexing mode from among a plurality of spatial multiplexing modes, wherein each of the plurality of spatial multiplexing modes supports simultaneous transmission of multiple data symbol streams via multiple spatial channels of a MIMO channel;

means for spatially processing a first plurality of data symbol streams in accordance with the selected spatial multiplexing mode to obtain a plurality of transmit symbol streams;

means for transmitting the plurality of transmit symbol streams from a plurality of antennas and via a first communication link;

means for receiving a plurality of received symbol streams from the plurality of antennas for a second communication link; and

means for spatially processing the plurality of received symbol streams in accordance with the selected spatial multiplexing mode to obtain a plurality of recovered data symbol streams, which are estimates of a second plurality of data symbol streams sent via the second communication link.

26. The apparatus of claim 25, wherein the plurality of spatial multiplexing modes include a steered mode and a non-steered mode, the steered mode supporting simultaneous transmission of multiple data symbol streams via multiple orthogonal spatial channels of the MIMO channel, and the non-steered mode supporting simultaneous transmission of multiple data symbol streams from the plurality of antennas.

27. The apparatus of claim 26, wherein the first plurality of data symbol streams are multiplied with a matrix of steering vectors for the steered mode and with an identity matrix for the non-steered mode, and wherein the plurality of received symbol streams are multiplied with a matrix of eigenvectors for the steered mode and with a spatial filter matrix for the non-steered mode.

28. The apparatus of claim 27, further comprising:
means for estimating a channel response of the second communication link; and
means for deriving the spatial filter matrix based on the estimated channel response for the second communication link.

29. The apparatus of claim 28, wherein the spatial filter matrix is derived based on a channel correlation matrix inversion (CCMI) technique, a minimum mean square error (MMSE) technique, or a successive interference cancellation (SIC) technique.

30. An access point in a wireless multiple-input multiple-output (MIMO) communication system, comprising:

a mode selector operable to select a spatial multiplexing mode from among a plurality of spatial multiplexing modes supported by the access point, wherein each of the plurality of spatial multiplexing modes supports simultaneous transmission of multiple data symbol streams via multiple spatial channels of a MIMO channel formed with a plurality of antennas at the access point;

a transmit spatial processor operable to spatially process a first plurality of data symbol streams in accordance with the selected spatial multiplexing mode to obtain a plurality of transmit symbol streams for transmission from the plurality of antennas and via a first communication link; and

a receive spatial processor operable to spatially process a plurality of received symbol streams, obtained from the plurality of antennas, in accordance with the selected spatial multiplexing mode to obtain a plurality of recovered data symbol streams, which are estimates of a second plurality of data symbol streams sent via a second communication link.

31. The access point of claim 30, wherein the plurality of spatial multiplexing modes include a steered mode and a non-steered mode.

32. The access point of claim 31, wherein

the transmit spatial processor is operable to multiply the first plurality of data symbol streams with a matrix of steering vectors for the steered mode and with an identity matrix for the non-steered mode, and

the receive spatial processor is operable to multiply the plurality of received symbol streams with a matrix of eigenvectors for the steered mode and with a spatial filter matrix for the non-steered mode.

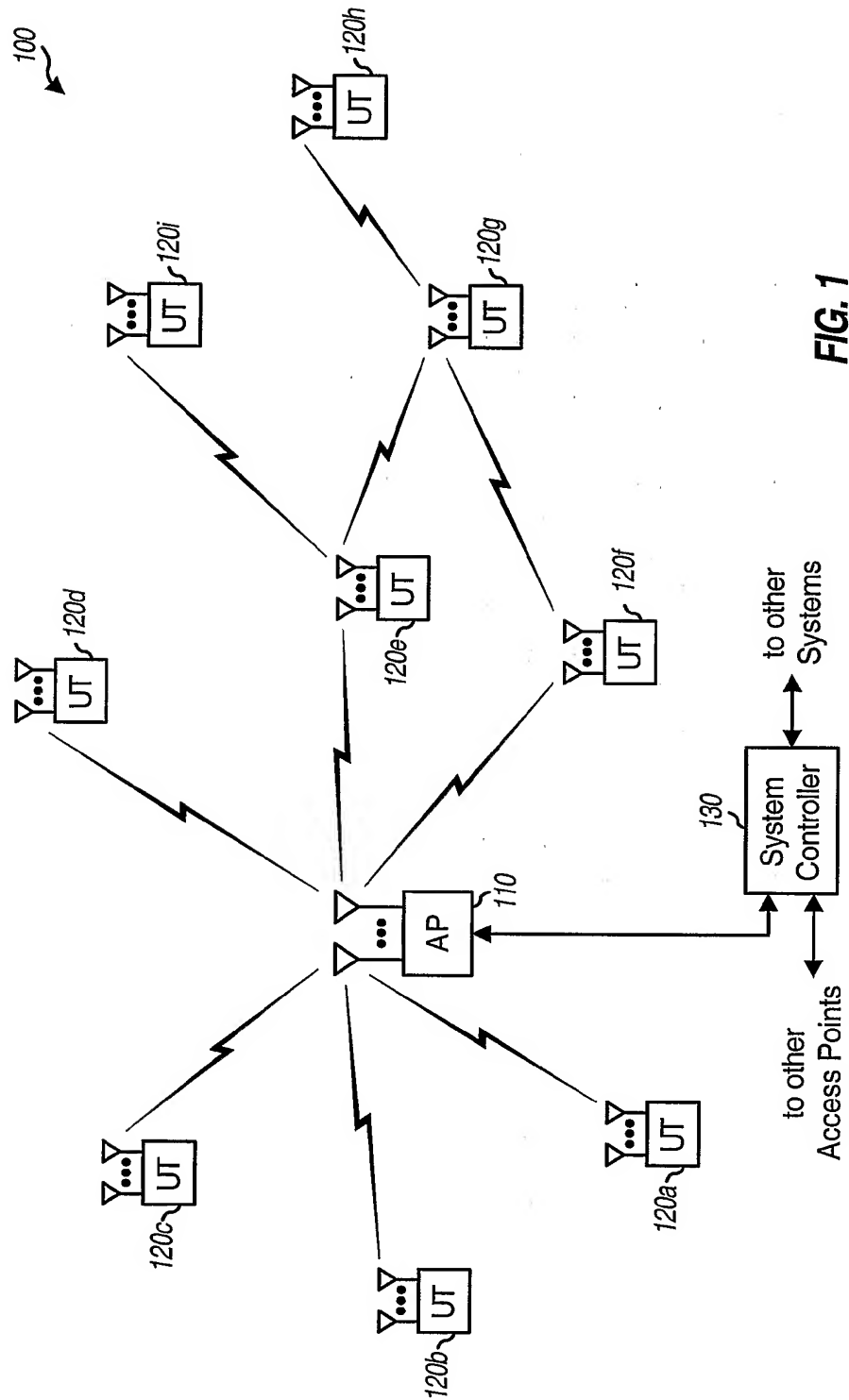


FIG. 1

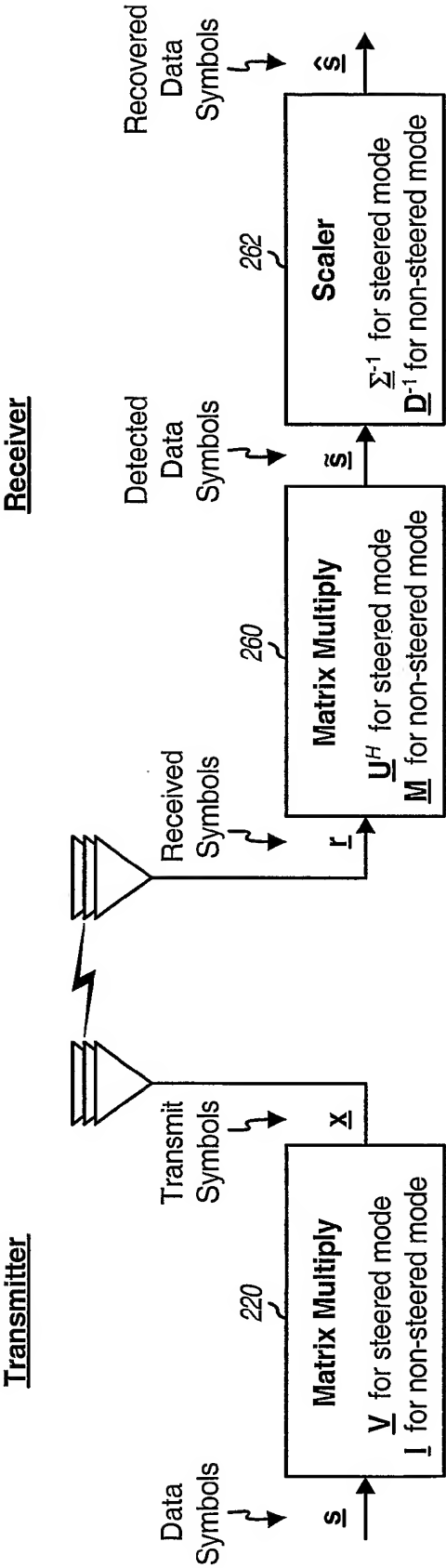


FIG. 2

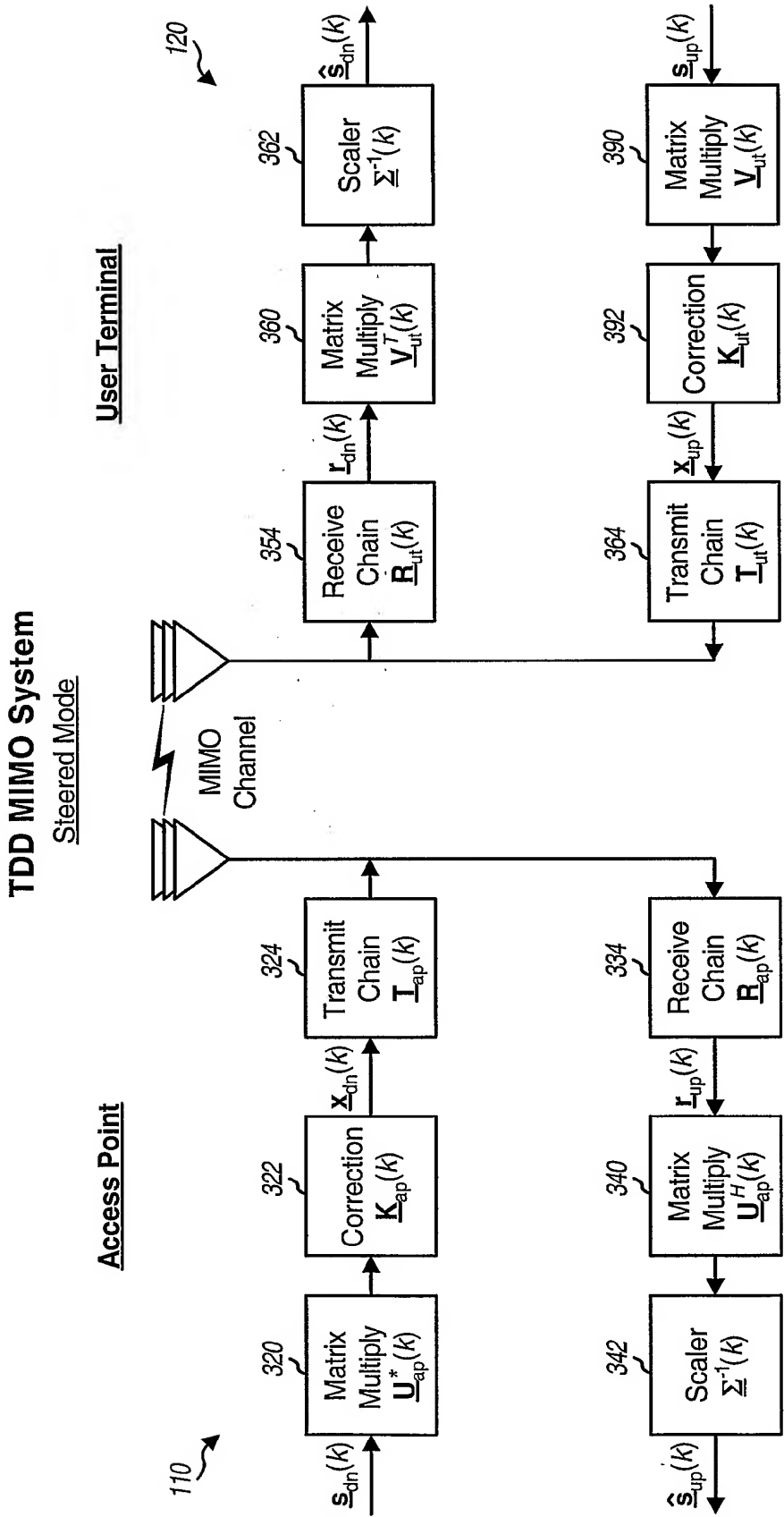


FIG. 3

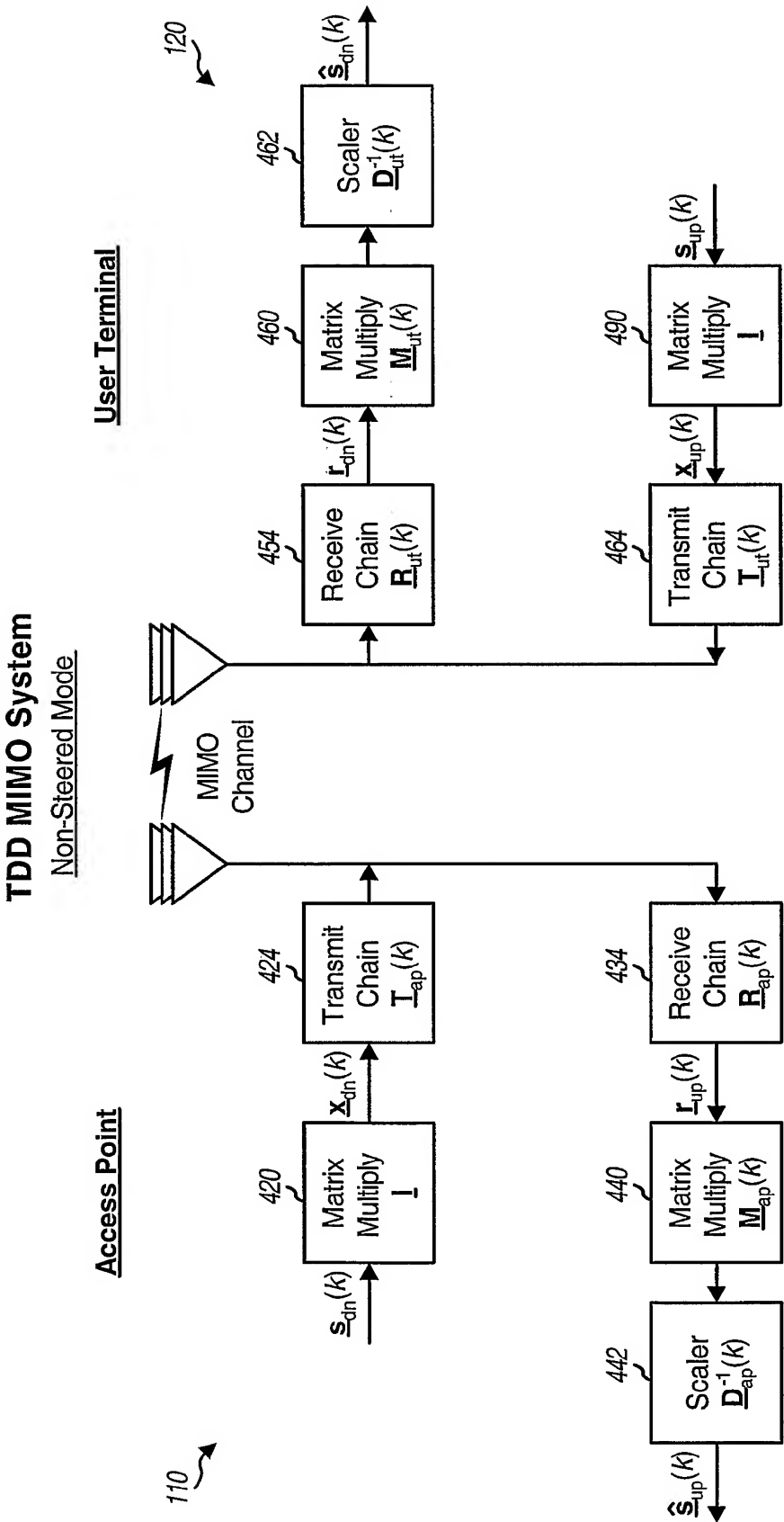


FIG. 4

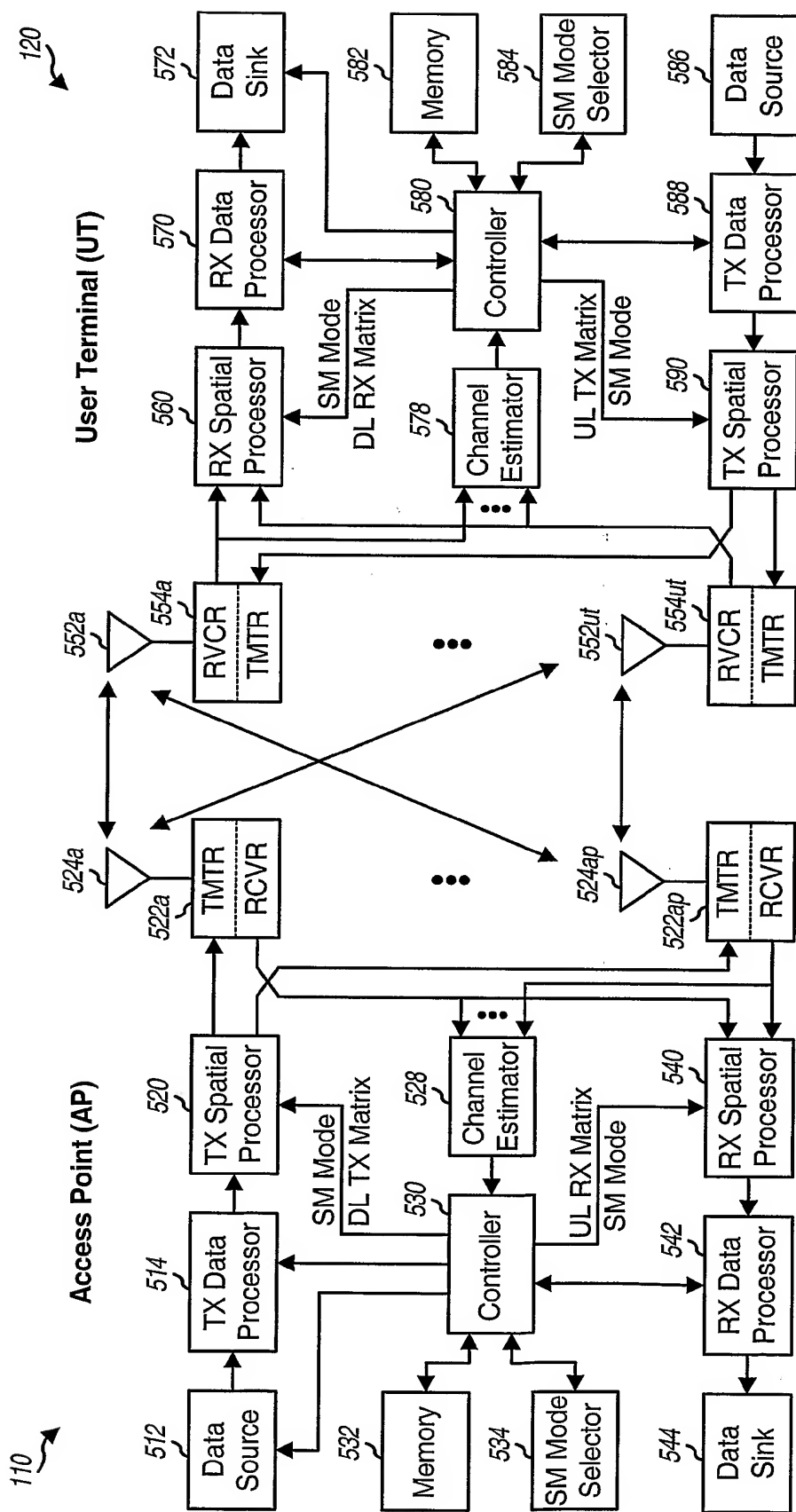
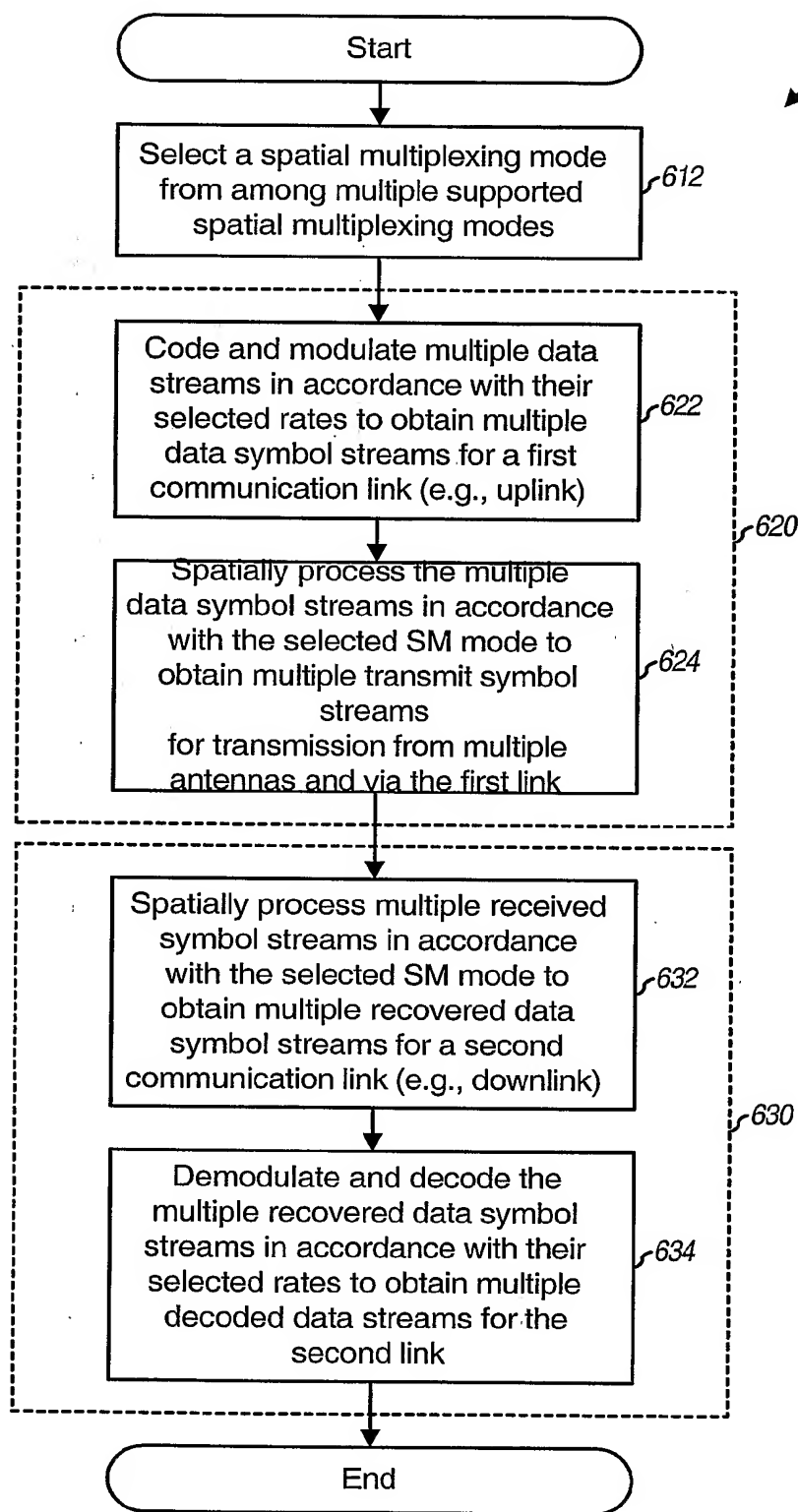


FIG. 5

**FIG. 6**

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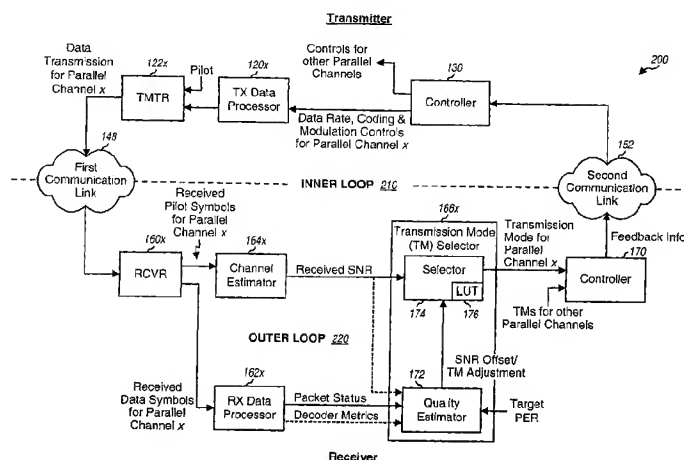
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(54) Title: CLOSED-LOOP RATE CONTROL FOR A MULTI-CHANNEL COMMUNICATION SYSTEM



(57) Abstract: Closed-loop rate control for data transmission on multiple parallel channels is provided. An inner loop estimates the channel conditions for a communication link and selects a suitable data rate for each of the multiple parallel channels based on the channel estimates. For each parallel channel, a received SNR is computed based on the channel estimates, an operating SNR is computed based on the received SNR and an SNR offset for the parallel channel, and the data rate is selected based on the operating SNR for the parallel channel and a set of required SNRs for a set of data rates supported by the system. An outer loop estimates the quality of data transmissions received on the multiple parallel channels and adjusts the operation of the inner loop. For example, the SNR offset for each parallel channel is adjusted based on the status of packets received on that parallel channel.

CLOSED-LOOP RATE CONTROL FOR A MULTI-CHANNEL COMMUNICATION SYSTEM

BACKGROUND

I. Field

[1001] The present invention relates generally to data communication, and more specifically to techniques for performing rate control for data transmission on multiple parallel channels in a multi-channel communication system.

II. Background

[1002] A multi-channel communication system utilizes multiple “parallel channels” for data transmission. These parallel channels may be formed in the time domain, frequency domain, spatial domain, or a combination thereof. For example, the multiple parallel channels may be formed by different time slots in a time division multiplex (TDM) communication system, different frequency subbands in a frequency division multiplex (FDM) communication system, different disjoint sets of subbands in an orthogonal frequency division multiplex (OFDM) communication system, or different spatial channels in a multiple-input multiple-output (MIMO) communication system. TDM, FDM, OFDM, and MIMO systems are described in further detail below.

[1003] The multiple parallel channels may experience different channel conditions (e.g., different fading, multipath, and interference effects) and may achieve different signal-to-noise ratios (SNRs). The SNR of a parallel channel determines its transmission capability, which is typically quantified by a particular data rate that may be reliably transmitted on the parallel channel. If the SNR varies from parallel channel to parallel channel, then the supported data rate would also vary from channel to channel. Moreover, since the channel conditions typically vary with time, the data rates supported by the multiple parallel channels also vary with time.

[1004] Rate control is a major challenge in a multi-channel communication system that experiences continually varying channel conditions. Rate control entails controlling the data rate of each of the multiple parallel channels based on the channel conditions. The goal of the rate control should be to maximize the overall throughput

on the multiple parallel channels while meeting certain quality objectives, which may be quantified by a particular packet error rate (PER) or some other criterion.

[1005] There is therefore a need in the art for techniques to effectively perform rate control for multiple parallel channels having varying SNRs.

SUMMARY

[1006] Techniques for performing closed-loop rate control for data transmission on multiple parallel channels are described herein. Closed-loop rate control may be achieved with one or multiple loops. An inner loop estimates the channel conditions for a communication link and selects a suitable data rate for each of the multiple parallel channels (e.g., to achieve high overall throughput). An outer loop (which is optional) estimates the quality of the data transmissions received on the multiple parallel channels and adjusts the operation of the inner loop.

[1007] For the inner loop, channel estimates are initially obtained for the multiple parallel channels (e.g., based on received pilot symbols). The channel estimates may include channel gain estimates for multiple subbands of each parallel channel, an estimate of the noise floor at the receiver, and so on. A suitable "transmission mode" is then selected for each parallel channel based on (1) the transmit power allocated to the parallel channel, (2) the channel estimates for the parallel channel, (3) an SNR offset provided by the outer loop for the parallel channel, and (4) other information provided by the outer loop. A transmission mode indicates, among other things, a specific data rate to use for a parallel channel. The SNR offset indicates the amount of back-off to use for the parallel channel and influences the selection of the transmission mode for the parallel channel. The other information from the outer loop may direct the inner loop to select a transmission mode with a data rate lower than that normally selected for the parallel channel, for example, if excessive packet errors are received for the parallel channel. The transmitter and receiver process data for each parallel channel in accordance with the transmission mode selected for that parallel channel.

[1008] For the outer loop, the receiver estimates the quality of the data transmissions received via the multiple parallel channels. For example, the receiver may determine the status of each received data packet (e.g., as good or bad, as described below), obtain decoder metrics for each data stream, estimate the received SNR for each parallel channel, and so on. The outer loop then adjusts the operation of the inner loop

for each parallel channel based on the estimated received quality for that parallel channel. For example, the outer loop may adjust the SNR offset for each parallel channel to achieve a target packet error rate (PER) for that parallel channel. The outer loop may also direct the inner loop to select a transmission mode with a lower data rate for a parallel channel if excessive packet errors are detected for that parallel channel.

[1009] Various aspects and embodiments of the invention are also described in further detail below.

BRIEF DESCRIPTION OF THE DRAWINGS

[1010] The features, nature, and advantages of the present invention will become more apparent from the detailed description set forth below when taken in conjunction with the drawings in which like reference characters identify correspondingly throughout and wherein:

[1011] FIG. 1 shows a transmitter and a receiver in a multi-channel communication system with closed-loop rate control for N_C parallel channels;

[1012] FIG. 2 shows a closed-loop rate control mechanism;

[1013] FIG. 3 shows an exemplary process to transmit N_C data streams on N_C parallel channels using N_C transmission modes selected with closed-loop rate control;

[1014] FIG. 4 shows an exemplary process for the outer loop;

[1015] FIG. 5 shows an exemplary TDD MIMO-OFDM system;

[1016] FIG. 6 shows a frame structure used in the TDD MIMO-OFDM system;

[1017] FIG. 7 shows a process for transmitting multiple data streams on multiple wideband eigenmodes on the downlink and uplink in the TDD MIMO-OFDM system;

[1018] FIG. 8 shows a process for selecting N_S transmission modes for N_S wideband eigenmodes;

[1019] FIGS. 9A and 9B show an access point and a terminal in the TDD MIMO-OFDM system for downlink and uplink transmission, respectively;

[1020] FIG. 10 shows a transmitter subsystem;

[1021] FIG. 11 shows a receiver subsystem; and

[1022] FIGS. 12A and 12B show exemplary timing diagrams for closed-loop rate control for the downlink and uplink, respectively.

DETAILED DESCRIPTION

[1023] The word “exemplary” is used herein to mean “serving as an example, instance, or illustration.” Any embodiment or design described herein as “exemplary” is not necessarily to be construed as preferred or advantageous over other embodiments or designs.

[1024] As used herein, “rate control” entails controlling the data rate of each of multiple parallel channels based on channel conditions. The data rate for each parallel channel is determined by the transmission mode selected for use for that parallel channel. Rate control may thus be achieved by controlling the transmission modes used for the multiple parallel channels.

[1025] FIG. 1 shows a block diagram of a transmitter 110 and a receiver 150 in a multi-channel communication system 100 with closed-loop rate control for N_C parallel channels, where $N_C > 1$. The N_C parallel channels may be formed in various manners, as described below. For downlink transmission, transmitter 110 is an access point, receiver 150 is a user terminal, first communication link 148 is the downlink (i.e., forward link), and second communication link 152 is the uplink (i.e., reverse link). For uplink transmission, transmitter 110 is a user terminal, receiver 150 is an access point, and the first and second communication links are the uplink and downlink, respectively.

[1026] At transmitter 110, a transmit (TX) data processor 120 receives N_C data streams, one stream for each of the N_C parallel channels. Each parallel channel is associated with a specific transmission mode that indicates a set of transmission parameters to use for that parallel channel. A transmission mode may indicate (or may be associated with) a particular data rate, a particular coding scheme or code rate, a particular interleaving scheme, a particular modulation scheme, and so on, to use for data transmission. An exemplary set of transmission modes is given in Table 2 below. For each parallel channel, the data rate is indicated by a data rate control, the coding scheme is indicated by a coding control, and the modulation scheme is indicated by a modulation control. These controls are provided by a controller 130 and are generated based on the transmission mode selected for each parallel channel using feedback information obtained from receiver 150 and possibly other information (e.g., channel estimates) obtained by transmitter 110.

[1027] TX data processor 120 codes, interleaves, and modulates each data stream in accordance with the transmission mode selected for its parallel channel to provide a

corresponding stream of modulation symbols. TX data processor 120 provides N_C modulation symbol streams for the N_C data streams. A transmitter unit (TMTR) 122 then processes the N_C modulation symbol streams in a manner specified by the system. For example, transmitter unit 122 may perform OFDM processing for an OFDM system, spatial processing for a MIMO system, or both spatial and OFDM processing for a MIMO-OFDM system (which is a MIMO system that utilizes OFDM). A pilot is also transmitted to assist receiver 150 in performing a number of functions such as channel estimation, acquisition, frequency and timing synchronization, coherent demodulation, and so on. Transmitter unit 122 multiplexes pilot symbols with the modulation symbols for each parallel channel, processes the multiplexed symbols, and provides a modulated signal for each antenna used for data transmission. Each modulated signal is then transmitted via first communication link 148 to receiver 150. First communication link 148 distorts each modulated signal with a particular channel response and further degrades the modulated signal with (1) additive white Gaussian noise (AWGN) having a variance of N_0 and (2) possibly interference from other transmitters.

[1028] At receiver 150, the transmitted signal(s) are received by one or more receive antennas, and the received signal from each antenna is provided to a receiver unit (RCVR) 160. Receiver unit 160 conditions and digitizes each received signal to provide a corresponding stream of samples. Receiver unit 160 further processes the samples in a manner that is complementary to that performed by transmitter unit 122 to provide N_C streams of “recovered” symbols, which are estimates of the N_C streams of modulation symbols sent by transmitter 110.

[1029] A receive (RX) data processor 162 then processes the N_C recovered symbol streams in accordance with the N_C transmission modes selected for the N_C parallel channels to obtain N_C decoded data streams, which are estimates of the N_C data streams sent by transmitter 110. The processing by RX data processor 162 may include demodulation, deinterleaving, and decoding. RX data processor 162 may further provide the status of each received data packet and/or decoder metrics for each decoded data stream.

[1030] Receiver unit 160 also provides received pilot symbols for the N_C parallel channels to a channel estimator 164. Channel estimator 164 processes these received pilot symbols to obtain channel estimates for the N_C parallel channels. The channel

estimates may include, for example, channel gain estimates, noise variance N_0 estimate, and so on. The noise variance N_0 , which is the noise floor observed at receiver 150, includes channel noise, receiver circuitry noise, interference (i.e., cross-talk) from other transmitting entities, and so on.

[1031] A transmission mode (TM) selector 166 receives the channel estimates from channel estimator 164 and possibly packet status and/or decoder metrics from RX data processor 162. Transmission mode selector 166 computes an operating SNR for each of the N_C parallel channels based on the channel estimates and an SNR offset for that parallel channel. Transmission mode selector 166 then selects a suitable transmission mode for each parallel channel based on the operating SNR and outer loop information for the parallel channel. The transmission mode selection is described in detail below.

[1032] A controller 170 receives the N_C selected transmission modes, TM 1 through TM N_C , from transmission mode selector 166 and the packet status from RX data processor 162 (not shown). Controller 170 then assembles feedback information for transmitter 110. The feedback information may include the N_C selected transmission modes for the N_C parallel channels, acknowledgments (ACKs) and/or negative acknowledgments (NAKs) for received data packets, a pilot, and/or other information. The feedback information is then sent via second communication link 152 to transmitter 110. Transmitter 110 uses the feedback information to adjust the processing of the N_C data streams sent to receiver 150. For example, transmitter 110 may adjust the data rate, the coding scheme, the modulation scheme, or any combination thereof, for each of the N_C data streams sent on the N_C parallel channels to receiver 150. The feedback information is used to increase the efficiency of the system by allowing data to be transmitted at the best-known settings supported by first communication link 148.

[1033] In the embodiment shown in FIG. 1, the channel estimation and transmission mode selection are performed by receiver 150 and the N_C transmission modes selected for the N_C parallel channels are sent back to transmitter 110. In other embodiments, the channel estimation and transmission mode selection may be performed (1) by transmitter 110 based on feedback information obtained from receiver 150 and/or other information obtained by transmitter 110 or (2) jointly by both transmitter 110 and receiver 150.

[1034] FIG. 2 shows a block diagram of an embodiment of a closed-loop rate control mechanism 200, which includes an inner loop 210 that operates in conjunction

with an outer loop 220. For simplicity, the operation of inner loop 210 and outer loop 220 for only one parallel channel x is shown in FIG. 2. In general, the same processing may be performed independently for each of the N_C parallel channels.

[1035] For inner loop 210, channel estimator 164x estimates the channel conditions for parallel channel x and provides channel estimates (e.g., channel gain estimates and noise floor estimate). A selector 174 within transmission mode selector 166x computes a received SNR for parallel channel x based on (1) the channel estimates from channel estimator 164x and (2) an SNR offset and/or a transmission mode adjustment for parallel channel x from a quality estimator 172. For clarity, the received SNR is symbolically shown as being provided by channel estimator 164x to selector 174 in FIG. 2. Selector 174 then selects a transmission mode for parallel channel x based on the received information, as described below. The select transmission mode for parallel channel x is included in the feedback information sent by controller 170 to the transmitter. At the transmitter, controller 130 receives the selected transmission mode for parallel channel x and determines the data rate, coding, and modulation controls for parallel channel x . Data is then processed in accordance with these controls by TX data processor 120x, further multiplexed with pilot symbols and conditioned by transmitter unit 122x, and sent to the receiver. The channel estimation and transmission mode selection may be performed periodically, at scheduled times, whenever changes in the communication link are detected, only as necessary (e.g., prior to and during data transmission), or at other times.

[1036] Outer loop 220 estimates quality of the data transmission received on parallel channel x and adjusts the operation of inner loop 210 for parallel channel x . The received data symbols for parallel channel x are processed by RX data processor 162x, and the status of each received packet on parallel channel x and/or decoder metrics are provided to quality estimator 172. The decoder metrics may include a re-encoded symbol error rate (SER), a re-encoded power metric, a modified Yamamoto metric (for a convolutional decoder), minimum or average log-likelihood ratio (LLR) among bits in a decoded packet (for a Turbo decoder), and so on. The re-encoded SER is the error rate between the received symbols from receiver unit 160 and the re-encoded symbols obtained by processing (e.g., re-encoding, re-modulating, and so on) the decoded data from RX data processor 162. The modified Yamamoto metric is indicative of the confidence in the decoded data and is obtained based on the difference

between the selected (best) path through the trellis for the convolutional decoding and the next closest path through the trellis. The minimum or average LLR may also be used as an indication of the confidence of the decoded data. These decoder metrics, which are indicative of the quality of the data transmission received on parallel channel x , are known in the art.

[1037] Outer loop 220 can provide different types of information used to control the operation of inner loop 210. For example, outer loop 220 can provide an SNR offset for each parallel channel. The SNR offset is used in the computation of the operating SNR for the parallel channel, as described below. The operating SNR is then provided to a look-up table (LUT) 176 and used to select the transmission mode for the parallel channel. The SNR offset thus influences the selection of the transmission mode. Outer loop 220 can also provide a transmission mode adjustment for each parallel channel. This adjustment may direct inner loop 210 to select a transmission mode with a lower data rate for the parallel channel. The transmission mode adjustment directly impacts the selection of the transmission mode. The SNR offset and transmission mode adjustment are two mechanisms for controlling the operation of inner loop 210. Outer loop 220 may also be designed to provide other types of adjustments for inner loop 210. For simplicity, only the SNR offset and transmission mode adjustment are described below. Outer loop 220 may adjust the SNR offset and/or transmission mode in various manners, some of which are described below.

[1038] In a first embodiment, the SNR offset and/or transmission mode for each parallel channel are adjusted based on packet errors detected for the data stream received on that parallel channel. The data stream may be transmitted in packets, blocks, frames, or some other data units. (For simplicity, packet is used herein for the data unit.) Each packet may be coded with an error detection code (e.g., a cyclic redundancy check (CRC) code) that allows the receiver to determine whether the packet was decoded correctly or in error. Each parallel channel may be associated with a particular target packet error rate (PER) (e.g., 1% PER). Quality estimator 172 receives the status of each received packet and the target PER for parallel channel x and adjusts the SNR offset for parallel channel x accordingly. For example, the SNR offset for parallel channel x may be initialized to zero at the start of data transmission on parallel channel x . The SNR offset may thereafter be reduced by ΔDN for each good packet and increased by ΔUP for each bad packet, where ΔDN and ΔUP may be selected based on

the target PER and the desired response time for the outer loop. The SNR offset is typically a positive value or zero but may also be allowed to be a negative value (e.g., to account for a high initial estimate of the received SNR). Alternatively or additionally, quality estimator 172 may provide a directive to adjust the transmission mode for parallel channel x to the next lower data rate, for example, if a burst of packet errors is detected on parallel channel x . The SNR offset and/or transmission mode adjustment from quality estimator 172 are used by selector 174 to select the transmission mode for parallel channel x .

[1039] In a second embodiment, the SNR offset and/or transmission mode for each parallel channel are adjusted based on the decoder metrics for that parallel channel. The decoder metrics for each parallel channel can be used to estimate the quality of the data transmission received on that parallel channel. If a particular decoder metric for a given parallel channel is worse than a threshold selected for that metric, then the SNR offset and/or transmission mode for that parallel channel may be adjusted accordingly.

[1040] In a third embodiment, the SNR offset and/or transmission mode for each parallel channel are adjusted based on the received SNR and the required SNR for that parallel channel. The received SNR for each parallel channel may be determined based on the received pilot symbols for that parallel channel. The system may support a set of transmission modes (e.g., as shown in Table 2), and each supported transmission mode requires a different minimum SNR to achieve the target PER. Quality estimator 172 can determine an SNR margin for parallel channel x , which is the difference between the received SNR and the required SNR for parallel channel x . If the SNR margin for parallel channel x is a negative value, then the transmission mode for parallel channel x may be adjusted to the next lower data rate.

[1041] The third embodiment may also be used for a design whereby a packet is demultiplexed and transmitted across multiple parallel channels. If the packet is received in error, then it may not be possible to determine (just from the received packet) which one or ones of the parallel channels cause the packet to be received in error. If no other information is available, then it may be necessary to adjust the N_C SNR offsets and/or the N_C transmission modes for all N_C parallel channels, for example, so that the next lower data rate is used for each parallel channel. This may result in an excessive amount of reduction on the overall data rate. However, using the third embodiment, the parallel channel with the smallest SNR margin can be assumed to have

caused the packet error, and the transmission mode for this parallel channel can be adjusted to the next lower data rate.

[1042] The outer loop may also adjust the operation of the inner loop in other manners, and this is within the scope of the invention. In general, the outer loop operates at a rate that may be faster or slower than the rate of the inner loop. For example, the adjustment of the SNR offset by the outer loop may be dependent on many received packets. The outer loop can also adjust the data rate in between regularly scheduled inner loop calculations. Thus, depending on its specific design and manner of operation, the outer loop typically has more influence on the operation of the inner loop for longer data transmissions. For bursty transmissions, the outer loop may not have much or any influence on the operation of the inner loop.

[1043] FIG. 3 shows a flow diagram of a process 300 to transmit N_C data streams on N_C parallel channels using N_C transmission modes selected with closed-loop rate control. Process 300 may be implemented as shown in FIGS. 1 and 2. Initially, the receiver estimates the channel gains and the noise floor N_0 for the N_C parallel channels (step 312). The receiver then selects a transmission mode for each of the N_C parallel channels based on the channel gain estimates, the noise floor estimate, and outer loop information (if any) for that parallel channel (step 314). The outer loop information may include the SNR offset and/or transmission mode adjustment for each of the N_C parallel channels. The transmission mode selection is described below. The receiver sends the N_C selected transmission modes for the N_C parallel channels, as feedback information, to the transmitter (step 316).

[1044] The transmitter codes and modulates the N_C data streams in accordance with the N_C selected transmission modes (obtained from the receiver) to provide N_C modulation symbol streams (step 322). The transmitter then processes and transmits the N_C modulation symbol streams on the N_C parallel channels to the receiver (step 324).

[1045] The receiver processes the data transmissions received on the N_C parallel channels from the transmitter and obtains N_C recovered symbol streams (step 332). The receiver further processes the N_C recovered symbol streams in accordance with the N_C selected transmission modes to obtain N_C decoded data streams (step 334). The receiver also estimates the quality of the data transmission received on each of the N_C parallel channels, e.g., based on the packet status, decoder metrics, received SNRs, and so on (step 336). The receiver then provides outer loop information for each of the N_C parallel

channels based on the estimated quality for the data transmission received on that parallel channel (step 338). In FIG. 3, steps 312 through 324 may be considered as part of the inner loop, and steps 332 through 338 may be considered as part of the outer loop.

[1046] FIG. 4 shows a flow diagram of a process 400 that may be performed for the outer loop. The status of data packets received on each of the N_C parallel channels is obtained and used to adjust the SNR offset and/or transmission mode for that parallel channel (step 412). Decoder metrics for each of the N_C parallel channels may also be obtained and used to adjust the SNR offset and/or transmission mode for that parallel channel (step 414). The received SNR for each of the N_C parallel channels may also be obtained for each parallel channel and used to compute the SNR margin for that parallel channel. The SNR margins for the N_C parallel channels may be used to adjust the transmission modes for the parallel channels if packet errors are detected (step 416). An outer loop may implement any one or any combination of the steps shown in FIG. 4, depending on its specific design.

[1047] The closed-loop rate control techniques described herein may be used for various types of multi-channel communication systems having multiple parallel channels that may be used for data transmission. For example, these techniques may be used for TDM systems, FDM systems, OFDM-based systems, MIMO systems, MIMO systems that utilize OFDM (i.e., MIMO-OFDM systems), and so on.

[1048] A TDM system may transmit data in frames, each of which may be of a particular time duration. Each frame may include multiple (N_{TS}) slots that may be assigned different slot indices. N_C parallel channels may be formed by the N_{TS} slots in each frame, where $N_C \leq N_{TS}$. Each of the N_C parallel channels may include one or multiple slots. The N_C channels are considered “parallel” even though they are not transmitted simultaneously.

[1049] An FDM system may transmit data in (N_{SB}) frequency subbands, which may be arbitrarily spaced. N_C parallel channels may be formed by the N_{SB} subbands, where $N_C \leq N_{SB}$. Each of the N_C parallel channels may include one or multiple subbands.

[1050] An OFDM system uses OFDM to effectively partition the overall system bandwidth into multiple (N_F) orthogonal subbands, which may also be referred to as tones, bins, and frequency channels. Each subband is associated with a respective carrier that may be modulated with data. N_C parallel channels may be formed by the N_F

subbands, where $N_C \leq N_F$. The N_C parallel channels are formed by N_C disjoint sets of one or more subbands. The N_C sets are disjoint in that each of the N_F subbands is assigned to only one set (and thus to one parallel channel), if at all. An OFDM system may be considered as a specific type of FDM system.

[1051] A MIMO system employs multiple (N_T) transmit antennas and multiple (N_R) receive antennas for data transmission, and is denoted as an (N_T, N_R) system. A MIMO channel formed by the N_T transmit and N_R receive antennas is composed of N_S spatial channels that may be used for data transmission, where $N_S \leq \min \{N_T, N_R\}$. The number of spatial channels is determined by a channel response matrix $\underline{\mathbf{H}}$ that describes the response between the N_T transmit and N_R receive antennas. For simplicity, the following description assumes that the channel response matrix $\underline{\mathbf{H}}$ is full rank. In this case, the number of spatial channels is given as $N_S = N_T \leq N_R$. N_C parallel channels may be formed by the N_S spatial channels, where $N_C \leq N_S$. Each of the N_C parallel channels may include one or multiple spatial channels.

[1052] A MIMO-OFDM system has N_S spatial channels for each of N_F subbands. N_C parallel channels may be formed by the N_S spatial channels of each of the N_F subbands, where $N_C \leq N_F \cdot N_S$. Each of the N_C parallel channels may include one or multiple spatial channels of one or multiple subbands (i.e., any combination of spatial channels and subbands). For MIMO and MIMO-OFDM systems, N_C parallel channels may also be formed by the N_T transmit antennas, where $N_C \leq N_T$. Each of the N_C parallel channels may be associated with one or multiple transmit antennas for data transmission.

[1053] For MIMO and MIMO-OFDM systems, data may be transmitted on the N_S spatial channels in various manners. For a partial channel state information (partial-CSI) MIMO system, data is transmitted on the N_S spatial channels without any spatial processing at the transmitter and with spatial processing at the receiver. For a full-CSI MIMO system, data is transmitted on the N_S spatial channels with spatial processing at both the transmitter and the receiver. For the full-CSI MIMO system, eigenvalue decomposition or singular value decomposition may be performed on the channel response matrix $\underline{\mathbf{H}}$ to obtain N_S "eigenmodes" of the MIMO channel. Data is transmitted on the N_S eigenmodes, which are orthogonalized spatial channels.

[1054] The closed-loop rate control techniques described herein may be used for time division duplex (TDD) systems as well as frequency division duplex (FDD) systems. For a TDD system, the downlink and uplink share the same frequency band and are likely to observe similar fading and multipath effects. Thus, the channel response for each link may be estimated based on a pilot received on either that link or the other link. For an FDD system, the downlink and uplink use different frequency bands and are likely to observe different fading and multipath effects. The channel response for each link may be estimated based on a pilot received on that link.

[1055] The closed-loop rate control techniques may be used for both partial-CSI and full-CSI MIMO systems. These techniques may also be used for the downlink as well as the uplink.

[1056] The closed-loop rate control techniques are now described in detail for an exemplary multi-channel communication system, which is a full-CSI TDD MIMO-OFDM system. For simplicity, in the following description, the term “eigenmode” and “wideband eigenmode” are used to denote the case where an attempt is made to orthogonalize the spatial channels, even though it may not be fully successful due to, for example, an imperfect channel estimate.

I. TDD MIMO-OFDM System

[1057] FIG. 5 shows an exemplary TDD MIMO-OFDM system 500 with a number of access points (APs) 510 that support communication for a number of user terminals (UTs) 520. For simplicity, only two access points 510a and 510b are shown in FIG. 5. An access point may also be referred to as a base station, a base transceiver system, a Node B, or some other terminology. A user terminal may be fixed or mobile, and may also be referred to as an access terminal, a mobile station, a user equipment (UE), a wireless device, or some other terminology. Each user terminal may communicate with one or possibly multiple access points on the downlink and/or the uplink at any given moment. A system controller 530 couples to access points 510 and provides coordination and control for these access points.

[1058] FIG. 6 shows an exemplary frame structure 600 that may be used in TDD MIMO-OFDM system 500. Data transmission occurs in units of TDD frames, each of which spans a particular time duration (e.g., 2 msec). Each TDD frame is partitioned into a downlink phase and an uplink phase, and each phase is further partitioned into

multiple segments for multiple transport channels. In the embodiment shown in FIG. 6, the downlink transport channels include a broadcast channel (BCH), a forward control channel (FCCH), and a forward channel (FCH), and the uplink transport channels include a reverse channel (RCH) and a random access channel (RACH).

[1059] In the downlink phase, a BCH segment 610 is used to transmit one BCH protocol data unit (PDU) 612, which includes a beacon pilot 614, a MIMO pilot 616, and a BCH message 618. The beacon pilot is a pilot transmitted from all antennas and is used for timing and frequency acquisition. The MIMO pilot is a pilot transmitted from all antennas but with a different orthogonal code for each antenna in order to allow the user terminals to individually identify the antennas. The MIMO pilot is used for channel estimation. The BCH message carries system parameters for the user terminals. An FCCH segment 620 is used to transmit one FCCH PDU, which carries assignments for downlink and uplink resources (e.g., the selected transmission modes for the downlink and uplink) and other signaling for the user terminals. An FCH segment 630 is used to transmit one or more FCH PDUs 632 on the downlink. Different types of FCH PDU may be defined. For example, an FCH PDU 632a includes a steered reference 634a and a data packet 636a, and an FCH PDU 632b includes only a data packet 636b. The steered reference is a pilot that is transmitted on a specific wideband eigenmode (as described below) and is used for channel estimation.

[1060] In the uplink phase, an RCH segment 640 is used to transmit one or more RCH PDUs 642 on the uplink. Different types of RCH PDU may also be defined. For example, an RCH PDU 642a includes only a data packet 646a, and an RCH PDU 642b includes a steered reference 644b and a data packet 646b. An RACH segment 650 is used by the user terminals to gain access to the system and to send short messages on the uplink. An RACH PDU 652 may be sent in RACH segment 650 and includes a pilot (e.g., steered reference) 654 and a message 656.

[1061] FIG. 6 shows an exemplary frame structure for a TDD system. Other frame structures may also be used, and this is within the scope of the invention.

1. Spatial Processing

[1062] For a MIMO-OFDM system, the channel response between an access point and a user terminal may be characterized by a set of channel response matrices, $\underline{\mathbf{H}}(k)$ for $k \in K$, where K represents the set of all subbands of interest (e.g., $K = \{1, \dots, N_F\}$).

For a TDD MIMO-OFDM system with a shared frequency band, the downlink and uplink channel responses may be assumed to be reciprocal of one another. That is, if $\underline{\mathbf{H}}(k)$ represents a channel response matrix from antenna array A to antenna array B for subband k , then a reciprocal channel implies that the coupling from array B to array A is given by $\underline{\mathbf{H}}^T(k)$, where $\underline{\mathbf{A}}^T$ denotes the transpose of $\underline{\mathbf{A}}$.

[1063] However, the frequency responses of the transmit and receive chains at the access point are typically different from the frequency responses of the transmit and receive chains at the user terminal. Calibration may be performed to obtain correction matrices used to account for differences in the frequency responses. With these correction matrices, the “calibrated” downlink channel response, $\underline{\mathbf{H}}_{\text{cdn}}(k)$, observed by the user terminal is the transpose of the “calibrated” uplink channel response, $\underline{\mathbf{H}}_{\text{cup}}(k)$, observed by the access point, i.e., $\underline{\mathbf{H}}_{\text{cdn}}(k) = \underline{\mathbf{H}}_{\text{cup}}^T(k)$, for $k \in K$. For simplicity, the following description assumes that the downlink and uplink channel responses are calibrated and reciprocal of one another.

[1064] On the downlink, a MIMO pilot may be transmitted by the access point (e.g., in BCH segment 610) and used by the user terminal to obtain an estimate of the calibrated downlink channel response, $\hat{\underline{\mathbf{H}}}_{\text{cdn}}(k)$, for $k \in K$. The user terminal may estimate the calibrated uplink channel response as $\hat{\underline{\mathbf{H}}}_{\text{cup}}(k) = \hat{\underline{\mathbf{H}}}_{\text{cdn}}^T(k)$. The user terminal may perform singular value decomposition of $\hat{\underline{\mathbf{H}}}_{\text{cup}}(k)$, for each subband k , as follows:

$$\hat{\underline{\mathbf{H}}}_{\text{cup}}(k) = \hat{\underline{\mathbf{U}}}_{\text{ap}}(k) \hat{\underline{\Sigma}}(k) \hat{\underline{\mathbf{V}}}_{\text{ut}}^H(k) \quad , \text{ for } k \in K \quad , \quad \text{Eq (1)}$$

where $\hat{\underline{\mathbf{U}}}_{\text{ap}}(k)$ is an $(N_{\text{ap}} \times N_{\text{ap}})$ unitary matrix of left eigenvectors of $\hat{\underline{\mathbf{H}}}_{\text{cup}}(k)$;

$\hat{\underline{\Sigma}}(k)$ is an $(N_{\text{ap}} \times N_{\text{ut}})$ diagonal matrix of singular values of $\hat{\underline{\mathbf{H}}}_{\text{cup}}(k)$;

$\hat{\underline{\mathbf{V}}}_{\text{ut}}(k)$ is an $(N_{\text{ut}} \times N_{\text{ut}})$ unitary matrix of right eigenvectors of $\hat{\underline{\mathbf{H}}}_{\text{cup}}(k)$;

$\underline{\mathbf{A}}^H$ is the conjugate transpose of $\underline{\mathbf{A}}$;

N_{ap} is the number of antennas at the access point; and

N_{ut} is the number of antennas at the user terminal.

[1065] Similarly, the singular value decomposition of $\hat{\underline{\mathbf{H}}}_{\text{cdn}}(k)$ may be expressed as:

$$\hat{\mathbf{H}}_{\text{cdn}}(k) = \hat{\mathbf{V}}_{\text{ut}}^*(k) \hat{\mathbf{\Sigma}}(k) \hat{\mathbf{U}}_{\text{ap}}^T(k) \quad , \text{ for } k \in K \quad , \quad \text{Eq (2)}$$

where $\hat{\mathbf{V}}_{\text{ut}}^*(k)$ and $\hat{\mathbf{U}}_{\text{ap}}^*(k)$ are unitary matrices of left and right eigenvectors, respectively, of $\hat{\mathbf{H}}_{\text{cdn}}(k)$ and “*” denotes the complex conjugate. Singular value decomposition is described by Gilbert Strang in a book entitled “Linear Algebra and Its Applications,” Second Edition, Academic Press, 1980.

[1066] As shown in equations (1) and (2), the matrices of left and right eigenvectors for one link are the complex conjugate of the matrices of right and left eigenvectors, respectively, for the other link. The matrices $\hat{\mathbf{U}}_{\text{ap}}(k)$ and $\hat{\mathbf{V}}_{\text{ut}}(k)$ may be used by the access point and the user terminal, respectively, for spatial processing and are denoted as such by their subscripts. The matrix $\hat{\mathbf{\Sigma}}(k)$ includes singular value estimates that represent the gains for the spatial channels (or eigenmodes) of the channel response matrix $\mathbf{H}(k)$ for each subband k .

[1067] Singular value decomposition may be performed independently for the channel response matrix $\hat{\mathbf{H}}_{\text{cup}}(k)$ for each subband k to determine the N_s eigenmodes of that subband. The singular value estimates for each diagonal matrix $\hat{\mathbf{\Sigma}}(k)$ may be ordered such that $\{\hat{\sigma}_1(k) \geq \hat{\sigma}_2(k) \geq \dots \geq \hat{\sigma}_{N_s}(k)\}$, where $\hat{\sigma}_1(k)$ is the largest singular value estimate and $\hat{\sigma}_{N_s}(k)$ is the smallest singular value estimate for subband k . When the singular value estimates for each diagonal matrix $\hat{\mathbf{\Sigma}}(k)$ are ordered, the eigenvectors (or columns) of the associated matrices $\hat{\mathbf{U}}(k)$ and $\hat{\mathbf{V}}(k)$ are also ordered correspondingly. A “wideband eigenmode” may be defined as the set of same-order eigenmodes of all subbands after the ordering. Thus, the m -th wideband eigenmode includes the m -th eigenmodes of all subbands. The “principal” wideband eigenmode is the one associated with the largest singular value estimate in the matrix $\hat{\mathbf{\Sigma}}(k)$ for each of the subbands. N_s parallel channels may be formed by the N_s wideband eigenmodes.

[1068] The user terminal may transmit a steered reference on the uplink (e.g., in RCH segment 640 or RACH segment 650 in FIG. 6). The uplink steered reference for wideband eigenmode m may be expressed as:

$$\mathbf{x}_{\text{up,st},m}(k) = \hat{\mathbf{v}}_{\text{ut},m}(k) p(k) \quad , \text{ for } k \in K \quad , \quad \text{Eq (3)}$$

where $\underline{\mathbf{x}}_{\text{up},\text{sr},m}(k)$ is a vector of N_{ut} symbols sent from N_{ut} user terminal antennas for subband k of wideband eigenmode m for the steered reference;

$\hat{\mathbf{y}}_{\text{ut},m}(k)$ is the m -th column of the matrix $\hat{\mathbf{V}}_{\text{ut}}(k)$ for subband k , where

$$\hat{\mathbf{V}}_{\text{ut}}(k) = [\hat{\mathbf{y}}_{\text{ut},1}(k) \ \hat{\mathbf{y}}_{\text{ut},2}(k) \ \dots \ \hat{\mathbf{y}}_{\text{ut},N_{\text{ut}}}(k)] ; \text{ and}$$

$p(k)$ is the pilot symbol sent on subband k .

The steered reference for all N_S wideband eigenmodes may be transmitted in N_S OFDM symbol periods, or fewer than N_S OFDM symbol periods using subband multiplexing. The steered reference for each wideband eigenmode may also be transmitted over multiple OFDM symbol periods.

[1069] The received uplink steered reference at the access point may be expressed as:

$$\begin{aligned} \underline{\mathbf{r}}_{\text{up},\text{sr},m}(k) &= \underline{\mathbf{H}}_{\text{cup}}(k) \hat{\mathbf{y}}_{\text{ut},m}(k) p(k) + \underline{\mathbf{n}}_{\text{up}}(k) \\ &\approx \underline{\hat{\mathbf{u}}}_{\text{ap},m}(k) \hat{\sigma}_m(k) p(k) + \underline{\mathbf{n}}_{\text{up}}(k) \end{aligned} \quad , \text{ for } k \in K, \quad \text{Eq (4)}$$

where $\underline{\mathbf{r}}_{\text{up},\text{sr},m}(k)$ is a vector of N_{ap} symbols received on N_{ap} access point antennas for subband k of wideband eigenmode m for the steered reference;

$\underline{\hat{\mathbf{u}}}_{\text{ap},m}(k)$ is the m -th column of the matrix $\hat{\mathbf{U}}_{\text{ap}}(k)$ for subband k , where

$$\hat{\mathbf{U}}_{\text{ap}}(k) = [\underline{\hat{\mathbf{u}}}_{\text{ap},1}(k) \ \underline{\hat{\mathbf{u}}}_{\text{ap},2}(k) \ \dots \ \underline{\hat{\mathbf{u}}}_{\text{ap},N_{\text{ap}}}(k)] ;$$

$\hat{\sigma}_m(k)$ is the singular value estimate for subband k of wideband eigenmode m ,

i.e., the m -th diagonal element of the matrix $\hat{\Sigma}(k)$; and

$\underline{\mathbf{n}}_{\text{up}}(k)$ is additive white Gaussian noise (AWGN) for subband k on the uplink.

[1070] As shown in equation (4), at the access point, the received steered reference (in the absence of noise) is approximately $\underline{\hat{\mathbf{u}}}_{\text{ap},m}(k) \hat{\sigma}_m(k) p(k)$. The access point can thus obtain estimates of both $\underline{\hat{\mathbf{u}}}_{\text{ap},m}(k)$ and $\hat{\sigma}_m(k)$ for each subband k based on the received steered reference for that subband. The estimate of $\hat{\sigma}_m(k)$ for subband k of wideband eigenmode m , $\hat{\sigma}_m(k)$, may be expressed as:

$$\hat{\hat{\sigma}}_m(k) = \|\underline{\mathbf{r}}_{\text{up},\text{sr},m}(k)\|^2 = \sum_{i=1}^{N_{\text{ap}}} |r_{\text{up},\text{sr},m,i}(k)|^2, \text{ for } k \in K \text{ and } m \in M, \quad \text{Eq (5)}$$

where $\|\underline{\mathbf{a}}\|$ denotes the 2-norm of $\underline{\mathbf{a}}$;

$r_{\text{up},\text{sr},m,i}(k)$ is the i -th element of the vector $\underline{\mathbf{r}}_{\text{up},\text{sr},m}(k)$; and

M represents the set of all wideband eigenmodes of interest, e.g., $M = \{1, \dots, N_s\}$.

[1071] The estimate of $\underline{\hat{\mathbf{u}}}_{\text{ap},m}(k)$ for subband k of wideband eigenmode m , $\underline{\hat{\mathbf{u}}}_{\text{ap},m}(k)$, may be expressed as:

$$\underline{\hat{\mathbf{u}}}_{\text{ap},m}(k) = \underline{\mathbf{r}}_{\text{up},\text{sr},m}(k) / \hat{\hat{\sigma}}_m(k), \text{ for } k \in K \text{ and } m \in M. \quad \text{Eq (6)}$$

The double hat for $\underline{\hat{\mathbf{u}}}_{\text{ap},m}(k)$ and $\hat{\hat{\sigma}}_m(k)$ indicates that these are estimates of estimates, i.e., estimates obtained by the access point for the estimates $\underline{\hat{\mathbf{u}}}_{\text{ap},m}(k)$ and $\hat{\hat{\sigma}}_m(k)$ obtained by the user terminal. If the steered reference for each wideband eigenmode is transmitted over multiple OFDM symbol periods, then the access point can average the received steered reference for each wideband eigenmode to obtain more accurate estimates of $\underline{\hat{\mathbf{u}}}_{\text{ap},m}(k)$ and $\hat{\hat{\sigma}}_m(k)$.

[1072] Table 1 summarizes the spatial processing at the access point and the user terminal for data transmission and reception on multiple wideband eigenmodes.

Table 1

	Downlink	Uplink
Access Point	Transmit : $\underline{\mathbf{x}}_{\text{dn}}(k) = \hat{\hat{\mathbf{U}}}_{\text{ap}}^*(k) \underline{\mathbf{s}}_{\text{dn}}(k)$	Receive : $\hat{\hat{\mathbf{s}}}_{\text{up}}(k) = \hat{\hat{\Sigma}}^{-1}(k) \hat{\hat{\mathbf{U}}}_{\text{ap}}^H(k) \underline{\mathbf{r}}_{\text{up}}(k)$
User Terminal	Receive : $\hat{\hat{\mathbf{s}}}_{\text{dn}}(k) = \hat{\hat{\Sigma}}^{-1}(k) \hat{\hat{\mathbf{V}}}_{\text{ut}}^T(k) \underline{\mathbf{r}}_{\text{dn}}(k)$	Transmit : $\underline{\mathbf{x}}_{\text{up}}(k) = \hat{\hat{\mathbf{V}}}_{\text{ut}}(k) \underline{\mathbf{s}}_{\text{up}}(k)$

In Table 1, $\underline{\mathbf{s}}(k)$ is a “data” vector of modulation symbols (obtained from the symbol mapping at the transmitter), $\underline{\mathbf{x}}(k)$ is a “transmit” vector of transmit symbols (obtained after spatial processing at the transmitter), $\underline{\mathbf{r}}(k)$ is a “received” vector of received symbols (obtained after OFDM processing at the receiver), and $\hat{\hat{\mathbf{s}}}(k)$ is an estimate of

the vector $\underline{s}(k)$ (obtained after spatial processing at the receiver), where all of the vectors are for subband k . The subscripts “dn” and “up” for these vectors denote downlink and uplink, respectively. In Table 1, $\underline{\Sigma}^{-1}(k)$ is a diagonal matrix defined as $\underline{\Sigma}^{-1}(k) = \text{diag} (1/\sigma_1(k) \ 1/\sigma_2(k) \dots 1/\sigma_{N_S}(k))$.

[1073] The steered reference may be transmitted for one wideband eigenmode at a time by the user terminal or may be transmitted for multiple wideband eigenmodes simultaneously using an orthogonal basis (e.g., Walsh codes). The steered reference for each wideband eigenmode may be used by the access point to obtain $\hat{\underline{u}}_{\text{ap},m}(k)$, for $k \in K$, for that wideband eigenmode. If the N_S vectors $\hat{\underline{u}}_{\text{ap},m}(k)$ of the matrix $\hat{\underline{U}}_{\text{ap}}(k)$ are obtained individually (and over different OFDM symbol periods) for the N_S eigenmodes of each subband, then, due to noise and other sources of degradation in the wireless link, the N_S vectors $\hat{\underline{u}}_{\text{ap},m}(k)$ of the matrix $\hat{\underline{U}}_{\text{ap}}(k)$ for each subband k are not likely to be orthogonal to one another. In this case, the N_S vectors of the matrix $\hat{\underline{U}}_{\text{ap}}(k)$ for each subband k may be orthogonalized using QR factorization, polar decomposition, or some other techniques.

[1074] At the access point, a received SNR estimate for subband k of wideband eigenmode m , $\gamma_{\text{ap},m}(k)$, may be expressed as:

$$\gamma_{\text{ap},m}(k) = \frac{P_{\text{up},m}(k) \cdot \hat{\sigma}_m^2(k)}{N_{0,\text{ap}}} , \text{ for } k \in K \text{ and } m \in M , \quad \text{Eq (7)}$$

where $P_{\text{up},m}(k)$ is the transmit power used by the user terminal for subband k of wideband eigenmode m on the uplink; and $N_{0,\text{ap}}$ is the noise floor at the access point.

[1075] At the user terminal, a received SNR estimate for subband k of wideband eigenmode m , $\gamma_{\text{ut},m}(k)$, may be expressed as:

$$\gamma_{\text{ut},m}(k) = \frac{P_{\text{dn},m}(k) \cdot \hat{\sigma}_m^2(k)}{N_{0,\text{ut}}} , \text{ for } k \in K \text{ and } m \in M , \quad \text{Eq (8)}$$

where $P_{\text{dn},m}(k)$ is the transmit power used by the access point for subband k of wideband eigenmode m on the downlink; and

$N_{0,\text{ut}}$ is the noise floor at the user terminal.

As shown in equations (7) and (8), the received SNR for each subband of each wideband eigenmode, $\gamma_m(k)$, is dependent on the channel gain (which is $\hat{\sigma}_m(k)$ or $\hat{\hat{\sigma}}_m(k)$), the receiver noise floor N_0 , and the transmit power $P_m(k)$. The received SNR may be different for different subbands and eigenmodes.

[1076] FIG. 7 shows a flow diagram of a process 700 for transmitting multiple data streams on multiple wideband eigenmodes on the downlink and uplink in the exemplary TDD MIMO-OFDM system. Process 700 assumes that calibration has already been performed and that the downlink and uplink channel responses are transpose of one another, i.e., $\hat{\mathbf{H}}_{\text{cup}}(k) \approx \hat{\mathbf{H}}_{\text{cdn}}^T(k)$. For process 700, channel estimation is performed in block 710, transmission mode selection is performed in block 730, and data transmission/reception is performed in block 760.

[1077] For channel estimation, the access point transmits a MIMO pilot on the downlink (e.g., on the BCH) (step 712). The user terminal receives and processes the MIMO pilot to obtain an estimate of the calibrated downlink channel response, $\hat{\mathbf{H}}_{\text{cdn}}(k)$, for $k \in K$ (step 714). The user terminal then estimates the calibrated uplink channel response as $\hat{\mathbf{H}}_{\text{cup}}(k) = \hat{\mathbf{H}}_{\text{cdn}}^T(k)$ and performs singular value decomposition (SVD) of $\hat{\mathbf{H}}_{\text{cup}}(k)$ to obtain the matrices $\hat{\underline{\Sigma}}(k)$ and $\hat{\underline{\mathbf{V}}}_{\text{ut}}(k)$, for $k \in K$, as shown in equation (1) (step 716). The user terminal then transmits an uplink steered reference (e.g., on the RACH or the RCH) using the matrices $\hat{\underline{\mathbf{V}}}_{\text{ut}}(k)$, for $k \in K$, as shown in equation (3) (step 718). The access point receives and processes the uplink steered reference to obtain the matrices $\hat{\underline{\Sigma}}(k)$ and $\hat{\underline{\mathbf{U}}}_{\text{ap}}(k)$, for $k \in K$, as described above (step 720).

[1078] For downlink data transmission, the user terminal selects a transmission mode (with the highest supported data rate) for each wideband eigenmode on the downlink based on the diagonal matrix $\hat{\underline{\Sigma}}(k)$, the noise floor $N_{0,\text{ut}}$ at the user terminal, and downlink outer loop information (e.g., SNR offsets and/or transmission mode adjustments for the downlink) (step 740). The transmission mode selection is described

below. The user terminal then sends feedback information, which includes the N_S transmission modes selected by the user terminal for the downlink and may further include the noise floor $N_{0,ut}$ at the user terminal (step 742). (The steered reference transmitted in step 718 may also be viewed as feedback information sent by the user terminal.)

[1079] For uplink data transmission, the access point selects N_S transmission modes for the N_S wideband eigenmodes on the uplink based on the diagonal matrix $\hat{\underline{\Sigma}}(k)$, the noise floor $N_{0,ap}$ at the access point, and uplink outer loop information (e.g., SNR offsets and/or transmission mode adjustments for the uplink) (step 750). The access point further selects the N_S transmission modes for the N_S wideband eigenmodes on the downlink based on the feedback information received from the user terminal (step 752). The access point then sends the selected transmission modes for both the downlink and uplink (e.g., on the FCCH) (step 754). The user terminal receives the selected transmission modes for both links (step 756).

[1080] For downlink data transmission, the access point (1) codes and modulates the data for each downlink wideband eigenmode in accordance with the transmission mode selected for that wideband eigenmode, (2) spatially processes the data vector $\underline{s}_{dn}(k)$ with the matrix $\hat{\underline{U}}_{ap}^*(k)$, as shown in Table 1, to obtain the transmit vector $\underline{x}_{dn}(k)$, for $k \in K$, and (3) transmits the vector $\underline{x}_{dn}(k)$ on the downlink (step 762). The user terminal (1) receives the downlink transmission, (2) performs matched filtering on the received vector $\underline{r}_{dn}(k)$ with $\hat{\underline{\Sigma}}^{-1}(k)\hat{\underline{V}}_{ut}^T(k)$, as also shown in Table 1, to obtain the vector $\hat{\underline{s}}_{dn}(k)$, for $k \in K$, and (3) demodulates and decodes the recovered symbols in accordance with the transmission mode selected for each downlink wideband eigenmode (step 764).

[1081] For uplink data transmission, the user terminal (1) codes and modulates the data for each uplink wideband eigenmode in accordance with the transmission mode selected for that wideband eigenmode, (2) spatially processes the data vector $\underline{s}_{up}(k)$ with the matrix $\hat{\underline{V}}_{ut}(k)$ to obtain the transmit vector $\underline{x}_{up}(k)$, for $k \in K$, and (3) transmits the vector $\underline{x}_{up}(k)$ on the uplink (step 772). The access point (1) receives the uplink transmission, (2) performs matched filtering on the received vector $\underline{r}_{up}(k)$ with

$\hat{\underline{\Sigma}}^{-1}(k)\hat{\underline{\mathbf{U}}}_{\text{ap}}^H(k)$ to obtain the vector $\hat{\underline{\mathbf{s}}}_{\text{up}}(k)$, and (3) demodulates and decodes the recovered symbols in accordance with the transmission mode selected for each uplink wideband eigenmode (step 774). For simplicity, the closed-loop operation and the transmission mode adjustment by the outer loop are not shown in FIG. 7.

[1082] FIG. 7 shows a specific embodiment of a process that may be used for downlink and uplink data transmission in the exemplary TDD MIMO-OFDM system. Other processes may also be implemented whereby the channel estimation, transmission mode selection, and/or data transmission/reception may be performed in some other manners.

2. Transmission Mode Selection

[1083] FIG. 8 shows a flow diagram of a process 800 for selecting N_S transmission modes for the N_S wideband eigenmodes. Process 800 may be used for steps 740 and 750 in FIG. 7. Initially, the total transmit power, P_{total} , available at the transmitter for data transmission is distributed to the N_S wideband eigenmodes based on a power distribution scheme (step 812). The transmit power P_m allocated to each wideband eigenmode is then distributed to the N_F subbands of that wideband eigenmode based on the same or a different power distribution scheme (step 814). The power distribution across the N_S wideband eigenmodes and the power distribution across the N_F subbands of each wideband eigenmode may be performed as described below.

[1084] An operating SNR for each wideband eigenmode, $\gamma_{\text{op},m}$, is computed based on (1) the allocated transmit powers $P_m(k)$ and the channel gains $\sigma_m(k)$ for the subbands of that wideband eigenmode, (2) the noise floor N_0 at the receiver, and (3) the SNR offset for that wideband eigenmode (step 816). The computation of the operating SNR is described below. A suitable transmission mode q_m is then selected for each wideband eigenmode based on the operating SNR for that wideband eigenmode and a look-up table (step 818). Excess power for each wideband eigenmode is determined, and the total excess power for all wideband eigenmodes is redistributed to one or more wideband eigenmodes to improve performance (step 820). The transmission mode for each wideband eigenmode may be adjusted (e.g., to the next lower data rate) if directed by outer loop information (step 822). Each of the steps in FIG. 8 is described in detail below.

A. Power Distribution Across Wideband Eigenmodes

[1085] For step 812 in FIG. 8, the total transmit power, P_{total} , may be distributed to the N_S wideband eigenmodes using various schemes. Some of these power distribution schemes are described below.

[1086] In a uniform power distribution scheme, the total transmit power, P_{total} , is distributed uniformly across the N_S wideband eigenmodes such that they are all allocated equal power. The transmit power P_m allocated to each wideband eigenmode m may be expressed as:

$$P_m = \frac{P_{\text{total}}}{N_S}, \text{ for } m \in M. \quad \text{Eq (9)}$$

[1087] In a water-filling power distribution scheme, the total transmit power, P_{total} , is distributed to the N_S wideband eigenmodes based on a “water-filling” or “water-pouring” procedure. The water-filling procedure distributes the total transmit power, P_{total} , across the N_S wideband eigenmodes such that the overall spectral efficiency is maximized. Water-filling is described by Robert G. Gallager in “Information Theory and Reliable Communication,” John Wiley and Sons, 1968. The water-filling for the N_S wideband eigenmodes may be performed in various manners, some of which are described below.

[1088] In a first embodiment, the total transmit power, P_{total} , is initially distributed to the $N_S N_F$ subbands/eigenmodes using water-filling and based on their received SNRs, $\gamma_m(k)$, for $k \in K$ and $m \in M$. The received SNR, $\gamma_m(k)$, may be computed as shown in equation (7) or (8) with the assumption of P_{total} being uniformly distributed across the $N_S N_F$ subbands/eigenmodes. The result of this power distribution is an initial transmit power, $P'_m(k)$, for each subband/eigenmode. The transmit power P_m allocated to each wideband eigenmode is then obtained by summing the initial transmit powers, $P'_m(k)$, allocated to the N_F subbands of that wideband eigenmode, as follows:

$$P_m = \sum_{k=1}^{N_F} P'_m(k), \text{ for } m \in M. \quad \text{Eq (10)}$$

[1089] In a second embodiment, the total transmit power, P_{total} , is distributed to the N_S wideband eigenmodes based on the average SNRs computed for these wideband eigenmodes. Initially, the average SNR, $\gamma_{\text{avg},m}$, is computed for each wideband eigenmode m based on the received SNRs for the N_F subbands of that wideband eigenmode, as follows:

$$\gamma_{\text{avg},m} = \frac{1}{N_F} \sum_{k=1}^{N_F} \gamma_m(k) , \quad \text{Eq (11)}$$

where $\gamma_m(k)$ is computed as described above for the first embodiment. Water-filling is then performed to distribute the total transmit power, P_{total} , across the N_S wideband eigenmodes based on their average SNRs, $\gamma_{\text{avg},m}$, for $m \in M$.

[1090] In a third embodiment, the total transmit power, P_{total} , is distributed to the N_S wideband eigenmodes based on the average SNRs for these wideband eigenmodes after channel inversion is applied for each wideband eigenmode. For this embodiment, the total transmit power, P_{total} , is first distributed uniformly to the N_S wideband eigenmodes. Channel inversion is then performed (as described below) independently for each wideband eigenmode to determine an initial power allocation, $P_m''(k)$, for each subband of that wideband eigenmode. After the channel inversion, the received SNR is the same across all subbands of each wideband eigenmode. The average SNR for each wideband eigenmode is then equal to the received SNR for any one of the subbands of that wideband eigenmode. The received SNR, $\gamma_m''(k)$, for one subband of each wideband eigenmode can be determined based on the initial power allocation, $P_m''(k)$, as shown in equation (7) or (8). The total transmit power, P_{total} , is then distributed to the N_S wideband eigenmodes using water-filling and based on their average SNRs, $\gamma_{\text{avg},m}''$, for $m \in M$.

[1091] Other schemes may also be used to distribute the total transmit power to the N_S wideband eigenmodes, and this is within the scope of the invention.

B. Power Allocation Across Subbands in Each Wideband Eigenmode

[1092] For step 814 in FIG. 8, the transmit power allocated to each wideband eigenmode, P_m , may be distributed to the N_F subbands of that wideband eigenmode using various schemes. Some of these power distribution schemes are described below.

[1093] In a uniform power distribution scheme, the transmit power for each wideband eigenmode, P_m , is distributed uniformly across the N_F subbands such that they are all allocated equal power. The transmit power $P_m(k)$ allocated to each subband may be expressed as:

$$P_m(k) = \frac{P_m}{N_F}, \text{ for } k \in K \text{ and } m \in M. \quad \text{Eq (12)}$$

For the uniform power distribution scheme, the received SNRs for the N_F subbands of each wideband eigenmode are likely to vary across the subbands.

[1094] In a channel inversion scheme, the transmit power for each wideband eigenmode, P_m , is distributed non-uniformly across the N_F subbands such that they achieve similar received SNRs at the receiver. In the following description, $\sigma_m(k)$ denotes the estimated channel gain, which is equal to $\hat{\sigma}_m(k)$ for the downlink and $\hat{\hat{\sigma}}_m(k)$ for the uplink. For the channel inversion scheme, a normalization b_m is initially computed for each wideband eigenmode, as follows:

$$b_m = \frac{1}{\sum_{k=1}^{N_F} [1 / \sigma_m^2(k)]}, \text{ for } m \in M. \quad \text{Eq (13)}$$

The transmit power $P_m(k)$ allocated to each subband of each wideband eigenmode may then be computed as:

$$P_m(k) = \frac{b_m \cdot P_m}{\sigma_m^2(k)}, \text{ for } k \in K \text{ and } m \in M. \quad \text{Eq (14)}$$

A transmit weight, $W_m(k)$, may be computed for each subband of each wideband eigenmode, as follows:

$$W_m(k) = \sqrt{P_m(k)}, \text{ for } k \in K \text{ and } m \in M. \quad \text{Eq (15)}$$

The transmit weights are used to scale modulation symbols at the transmitter. For the channel inversion scheme, all N_F subbands are used for each wideband eigenmode and the received SNRs for the subbands are approximately equal.

[1095] In a selective channel inversion scheme, the transmit power for each wideband eigenmode, P_m , is distributed non-uniformly across selected ones of the N_F subbands such that the selected subbands achieve similar received SNRs at the receiver. The selected subbands are those with channel gains equal to or greater than a gain threshold. For this scheme, an average power gain, g_m , is initially computed for each wideband eigenmode, as follows:

$$g_m = \frac{1}{N_F} \sum_{k=1}^{N_F} \sigma_m^2(k) , \text{ for } m \in M . \quad \text{Eq (16)}$$

A normalization \tilde{b}_m is then computed for each wideband eigenmode, as follows:

$$\tilde{b}_m = \frac{1}{\sum_{\sigma_m^2(k) > \beta_m g_m} [1 / \sigma_m^2(k)]} , \text{ for } m \in M , \quad \text{Eq (17)}$$

where $\beta_m g_m$ is the gain threshold and β_m is a scaling factor, which may be selected to maximize the overall throughput or based on some other criterion. The transmit power allocated to each subband of each wideband eigenmode, $P_m(k)$, may be expressed as:

$$P_m(k) = \begin{cases} \frac{\tilde{b}_m \cdot P_m}{\sigma_m^2(k)} , & \text{if } \sigma_m^2(k) \geq \beta_m g_m \\ 0 , & \text{otherwise} \end{cases} , \text{ for } k \in K \text{ and } m \in M . \quad \text{Eq (18)}$$

For the selective channel inversion scheme, N_F or fewer subbands may be selected for use for each wideband eigenmode and the received SNRs for the selected subbands are approximately equal.

[1096] Other schemes may also be used to distribute the transmit power P_m across the N_F subbands of each wideband eigenmode, and this is within the scope of the invention.

C. Transmission Mode Selection for Each Wideband Eigenmode

[1097] For step 816 in FIG. 8, an operating SNR is computed for each wideband eigenmode. The operating SNR indicates the transmission capability of the wideband eigenmode. Various methods may be used for step 816, depending on whether the received SNRs are similar or vary across the subbands of each wideband eigenmode. In the following description, SNRs are given in units of decibels (dB).

[1098] If channel inversion or selective channel inversion is performed, then the received SNRs for the subbands of each wideband eigenmode, $\gamma_m(k)$ for $k \in K$, are similar. The received SNR for subband k of wideband eigenmode m , $\gamma_m(k)$, may be computed as:

$$\gamma_m(k) = 10 \log_{10} \left(\frac{P_m(k) \cdot \sigma_m^2(k)}{N_0} \right), \text{ for } k \in K \text{ and } m \in M. \text{ (dB)} \quad \text{Eq (19)}$$

The operating SNR for each wideband eigenmode, $\gamma_{op,m}$, is equal to the received SNR for any one of the subbands of that wideband eigenmode minus the SNR offset for that wideband eigenmode, as follows:

$$\gamma_{op,m} = \gamma_m(k) - \gamma_{os,m}, \quad \text{for any } k \text{ and } m \in M, \text{ (dB)} \quad \text{Eq (20)}$$

where $\gamma_m(k)$, $\gamma_{os,m}$, and $\gamma_{op,m}$ are all given in units of dB in equations (19) and (20).

[1099] If the transmit power P_m for each wideband eigenmode is uniformly distributed across the subbands, then the received SNRs for the subbands of each wideband eigenmode are likely to vary. In this case, the operating SNR for each wideband eigenmode, $\gamma_{op,m}$, may be computed as:

$$\gamma_{op,m} = \gamma_{avg,m} - \gamma_{bo,m} - \gamma_{os,m}, \quad \text{(dB)} \quad \text{Eq (21)}$$

where $\gamma_{avg,m}$ is an average of the received SNRs for the N_F subbands of wideband eigenmode m ; and

$\gamma_{bo,m}$ is a back-off factor that accounts for variation in the received SNRs, which may be a function of the variance of the received SNRs.

[1100] For step 818 in FIG. 8, a suitable transmission mode is selected for each wideband eigenmode based on the operating SNR for that wideband eigenmode. The system may be designed to support a set of transmission modes. The transmission mode having index 0 is for a null data rate (i.e., no data transmission). Each supported transmission mode is associated with a particular minimum SNR required to achieve the desired level of performance (e.g., 1% PER). Table 2 lists an exemplary set of 14 transmission modes supported by the system, which are identified by transmission mode indices 0 through 13. Each transmission mode is associated with a particular spectral efficiency, a particular code rate, a particular modulation scheme, and the minimum SNR required to achieve 1% PER for a non-fading, AWGN channel. The spectral efficiency refers to the data rate (i.e., the information bit rate) normalized by the system bandwidth, and is given in units of bits per second per Hertz (bps/Hz). The spectral efficiency for each transmission mode is determined by the coding scheme and the modulation scheme for that transmission mode. The code rate and modulation scheme for each transmission mode in Table 2 are specific to the exemplary system design.

Table 2

Transmission Mode Index	Spectral Efficiency (bps/Hz)	Code Rate	Modulation Scheme	Required SNR (dB)
0	0.0	-	-	-
1	0.25	1/4	BPSK	-1.8
2	0.5	1/2	BPSK	1.2
3	1.0	1/2	QPSK	4.2
4	1.5	3/4	QPSK	6.8
5	2.0	1/2	16 QAM	10.1
6	2.5	5/8	16 QAM	11.7
7	3.0	3/4	16 QAM	13.2
8	3.5	7/12	64 QAM	16.2
9	4.0	2/3	64 QAM	17.4
10	4.5	3/4	64 QAM	18.8
11	5.0	5/6	64 QAM	20.0
12	6.0	3/4	256 QAM	24.2
13	7.0	7/8	256 QAM	26.3

[1101] For each supported transmission mode with a non-zero data rate, the required SNR is obtained based on the specific system design (i.e., the particular code rate, interleaving scheme, modulation scheme, and so on, used by the system for that transmission mode) and for an AWGN channel. The required SNR may be obtained by computer simulation, empirical measurements, and so on, as is known in the art. A look-up table may be used to store the set of supported transmission modes and their required SNRs.

[1102] The operating SNR for each wideband eigenmode, $\gamma_{op,m}$, may be provided to the look-up table, which then provides the transmission mode q_m for that wideband eigenmode. This transmission mode q_m is the supported transmission mode with the highest data rate and a required SNR, $\gamma_{req,m}$, that is less than or equal to the operating SNR (i.e., $\gamma_{req,m} \leq \gamma_{op,m}$). The look-up table thus selects the highest possible data rate for each wideband eigenmode based on the operating SNR for that wideband eigenmode.

D. Reallocation of Transmit Power

[1103] For step 820 in FIG. 8, the excess transmit power for each wideband eigenmode is determined and redistributed to improve performance. The following terms are used for the description below:

- Active wideband eigenmode - a wideband eigenmode with a non-zero data rate (i.e., a transmission mode having an index from 1 through 13 in Table 2);
- Saturated wideband eigenmode - a wideband eigenmode with the maximum data rate (i.e., transmission mode having index 13); and
- Unsaturated wideband eigenmode - an active wideband eigenmode with a non-zero data rate less than the maximum data rate (i.e., a transmission mode having an index from 1 through 12).

[1104] The operating SNR for a wideband eigenmode may be less than the smallest required SNR in the look-up table (i.e., $\gamma_{op,m} < -1.8$ dB for the transmission modes shown in Table 2). In this case, the wideband eigenmode may be shut off (i.e., not used) and the transmit power for this wideband eigenmode may be redistributed to other wideband eigenmodes.

[1105] The selected transmission mode q_m for each active wideband eigenmode is associated with a required SNR, $\gamma_{\text{req},m}$, that is equal to or lower than the operating SNR, i.e., $\gamma_{\text{req},m} \leq \gamma_{\text{op},m}$. The minimum transmit power required for each active wideband eigenmode, $P_{\text{req},m}$, may be computed as:

$$P_{\text{req},m} = \frac{P_m \cdot \gamma_{\text{req},m}}{\gamma_{\text{op},m}}, \text{ for } m \in M. \quad \text{Eq (22)}$$

The required transmit power is equal to zero ($P_{\text{req},m} = 0$) for each wideband eigenmode that is shut off (i.e., with transmission mode having index 0 in Table 2).

[1106] The excess power for each wideband eigenmode, $P_{\text{excess},m}$, is the amount of allocated power that is over the minimum power needed to achieve the required SNR (i.e., $P_{\text{excess},m} = P_m - P_{\text{req},m}$). The total excess power for all wideband eigenmodes, P_{excess} , may be computed as:

$$P_{\text{excess}} = \sum_{m=1}^{N_S} (P_m - P_{\text{req},m}) . \quad \text{Eq (23)}$$

[1107] The total excess power, P_{excess} , may be redistributed in various manners. For example, the total excess power, P_{excess} , may be redistributed to one or more wideband eigenmodes such that higher overall throughput is achieved. In one embodiment, the total excess power, P_{excess} , is redistributed to one unsaturated wideband eigenmode at a time, starting with the best one having the highest data rate, to move the wideband eigenmode to the next higher data rate. In another embodiment, the total excess power, P_{excess} , is redistributed to the wideband eigenmode that can achieve the highest increase in data rate with the least amount of transmit power.

[1108] If all wideband eigenmodes are operated at the highest data rate, or if the remaining excess power cannot increase the data rate of any wideband eigenmode, then the remaining excess power may be redistributed to one, multiple, or all active wideband eigenmodes to improve the SNR margins for these wideband eigenmodes.

E. Transmission Mode Adjustment

[1109] For step 822 in FIG. 8, the transmission mode for each wideband eigenmode may be adjusted based on information from the outer loop. The selected transmission modes for the downlink and uplink wideband eigenmodes may be adjusted using the techniques described above for FIG. 2. For example, if excessive packet errors are received on a given wideband eigenmode, then the outer loop may provide a transmission mode adjustment for that wideband eigenmode. As another example, a running average of the received SNRs may be maintained for each wideband eigenmode and used to compute the SNR margin for that wideband eigenmode. If the SNR margin for a given wideband eigenmode is negative, then the transmission mode for the wideband eigenmode may be adjusted to the next lower data rate. If a packet is transmitted across multiple wideband eigenmodes, then the transmission mode for the wideband eigenmode with the worse SNR margin may be adjusted to the next lower data rate whenever packet errors are detected. In any case, a transmission mode adjustment may direct the selection of another transmission mode with a lower data rate than the one selected in step 818.

II. MIMO-OFDM System

[1110] FIG. 9A shows a block diagram of an embodiment of an access point 510x and a user terminal 520x in the exemplary TDD MIMO-OFDM system. Access point 510x is one of access points 510 in FIG. 5, and user terminal 520x is one of user terminals 520. FIG. 9A shows the processing for downlink transmission. In this case, access point 510x is transmitter 110 in FIG. 1 and user terminal 520x is receiver 150.

[1111] For downlink transmission, at access point 510x, traffic data is provided from a data source 912 to a TX data processor 920, which demultiplexes the traffic data into N_C data streams, where $N_C > 1$. Traffic data may come from multiple data sources (e.g., one data source for each higher layer application) and the demultiplexing may not be needed. For simplicity, only one data source 912 is shown in FIG. 9A. TX data processor 920 formats, codes, interleaves, modulates, and scales each data stream in accordance with the transmission mode selected for that data stream to provide a corresponding scaled modulation symbol stream. The data rate, coding, and modulation for each data stream may be determined by a data rate control, a coding control, and a

modulation control, respectively, provided by a controller 940. TX data processor 920 provides N_C scaled modulation symbol streams to a TX spatial processor 928.

[1112] TX spatial processor 928 processes the N_C scaled modulation symbol streams based on a selected transmission scheme, multiplexes in pilot symbols, and provides N_{ap} transmit symbol streams to N_{ap} transmitter units (TMTR) 930a through 930ap. The selected transmission scheme may be for transmit diversity, spatial multiplexing, or beam-steering. Transmit diversity entails transmitting data redundantly from multiple antennas and/or on multiple subbands to obtain diversity and improve reliability. A space-time transmit diversity (STTD) may be used for transmit diversity. Beam-steering entails transmitting data on a single (best) spatial channel at full power using the phase steering information for the principal eigenmode. Spatial multiplexing entails transmitting data on multiple spatial channels to achieve higher spectral efficiency. The spatial processing for spatial multiplexing is shown in Table 1. Each transmitter unit 930 performs OFDM processing on its transmit symbol stream to provide a corresponding OFDM symbol stream, which is further processed to generate a modulated signal. The N_{ap} modulated signals from transmitter units 930a through 930ap are then transmitted via N_{ap} antennas 932a through 932ap, respectively.

[1113] At user terminal 520x, the N_{ap} transmitted signals are received by each of N_{ut} antennas 952a through 952ut, and the received signal from each antenna is provided to an associated receiver unit (RCVR) 954. Each receiver unit 954 conditions and digitizes its received signal to provide a stream of samples, which is further processed to provide a corresponding stream of received symbols. Receiver units 954a through 954ut provide N_{ut} received symbol streams to an RX spatial processor 962, which performs spatial processing based on the selected transmission scheme (e.g., as shown in Table 1 for spatial multiplexing). RX spatial processor 962 provides N_C recovered symbol streams, which are estimates of the N_C modulation symbol streams transmitted by access point 510x. An RX data processor 964 then demodulates, deinterleaves, and decodes each recovered symbol stream in accordance with the selected transmission mode to provide corresponding decoded data streams, which are estimates of the data streams transmitted by access point 510x. The processing by RX spatial processor 962 and RX data processor 964 is complementary to that performed by TX spatial processor 928 and TX data processor 920, respectively, at access point 510x.

[1114] A channel estimator 974 obtains estimates of one or more channel characteristics of the downlink and provides channel estimates to a controller 970. The channel estimates may be for channel gains, noise floor $N_{0,ut}$, and so on. RX data processor 964 may provide the status of each received data packet. Based on the various types of information received from channel estimator 974 and RX data processor 964, controller 970 determines a transmission mode for each of the multiple parallel channels on the downlink using the techniques described above. Each parallel channel may correspond to a wideband eigenmode (as described above) or some other combination of subbands and eigenmodes. Controller 970 provides feedback information, which may include the N_C selected transmission modes for the downlink, the channel estimates, the terminal noise floor, ACKs and/or NAKs for the receive data packets, and so on, or any combination thereof. The feedback information is processed by a TX data processor 978 and a TX spatial processor 980, multiplexed with a steered reference, conditioned by transmitter units 954a through 954ut, and transmitted via antennas 952a through 952ut to access point 510x.

[1115] At access point 510x, the N_{ut} transmitted signals from user terminal 520x are received by antennas 932a through 932ap, conditioned by receiver units 930a through 930ap, and processed by an RX spatial processor 934 and an RX data processor 936 to recover the feedback information sent by user terminal 520x. The feedback information is then provided to controller 940 and used to control the processing of the N_C data streams sent to user terminal 520x. For example, the data rate, coding, and modulation of each downlink data stream may be determined based on the transmission mode selected by user terminal 520x. The received ACK/NAK may be used to initiate either a full retransmission or an incremental transmission of each data packet received in error by user terminal 520x. For an incremental transmission, a small portion of a data packet received in error is transmitted to allow user terminal 520x to recover the packet.

[1116] A channel estimator 944 obtains channel gain estimates based on the received steered reference. The channel gain estimates are provided to controller 940 and used (possibly along with the user terminal noise floor $N_{0,ut}$ estimate) to derive transmit weights for the downlink. Controller 940 provides the data rate controls to data source 912 and TX data processor 920. Controller 940 further provides the coding and modulation controls and the transmit weights to TX data processor 920. The channel

estimation and transmission mode selection for downlink transmission may be performed as described above.

[1117] Controllers 940 and 970 direct the operation at access point 510x and user terminal 520x, respectively. Memory units 942 and 972 provide storage for program codes and data used by controllers 940 and 970, respectively.

[1118] FIG. 9B shows access point 510x and user terminal 520x for uplink transmission. In this case, user terminal 520x is transmitter 110 in FIG. 1 and access point 510x is receiver 150. The channel estimation and transmission mode selection for uplink transmission may be performed as described above. The data processing at access point 510x and user terminal 520x for uplink transmission may be performed in a manner similar to that described above for downlink transmission. The spatial processing at access point 510x and user terminal 520x for uplink transmission may be performed as shown in Table 1.

A. Transmitter and Receiver Subsystems

[1119] For clarity, the processing at access point 510x and user terminal 520x for downlink transmission is described in further detail below.

[1120] FIG. 10 shows a block diagram of a transmitter subsystem 1000, which is an embodiment of the transmitter portion of access point 510x. For this embodiment, TX data processor 920 includes a demultiplexer (Demux) 1010, N_C encoders 1012a through 1012s, N_C channel interleavers 1014a through 1014s, N_C symbol mapping units 1016a through 1016s, and N_C signal scaling units 1018a through 1018s (i.e., one set of encoder, channel interleaver, symbol mapping unit, and signal scaling unit for each of the N_C data streams). Demultiplexer 1010 demultiplexes the traffic data (i.e., the information bits) into N_C data streams, where each data stream is provided at the data rate indicated by the data rate control. Demultiplexer 1010 may be omitted if traffic data is already provided as N_C data streams.

[1121] Each encoder 1012 receives and codes a respective data stream based on the selected coding scheme (as indicated by the coding control) to provide code bits. Each data stream may carry one or more data packets, and each data packet is typically coded separately to obtain a coded data packet. The coding increases the reliability of the data transmission. The selected coding scheme may include any combination of CRC coding, convolutional coding, turbo coding, block coding, and so on. The code bits from each encoder 1012 are provided to a respective channel interleaver 1014, which

interleaves the code bits based on a particular interleaving scheme. If the interleaving is dependent on transmission mode, then controller 940 provides an interleaving control (as indicated by the dashed line) to channel interleaver 1014. The interleaving provides time, frequency, and/or spatial diversity for the code bits.

[1122] The interleaved bits from each channel interleaver 1014 are provided to a respective symbol mapping unit 1016, which maps the interleaved bits based on the selected modulation scheme (as indicated by the modulation control) to provide modulation symbols. Unit 1016 groups each set of B interleaved bits to form a B -bit binary value, where $B \geq 1$, and further maps each B -bit value to a specific modulation symbol based on the selected modulation scheme (e.g., QPSK, M-PSK, or M-QAM, where $M = 2^B$). Each modulation symbol is a complex value in a signal constellation defined by the selected modulation scheme. The modulation symbols from each symbol mapping unit 1016 are then provided to a respective signal scaling unit 1018, which scales the modulation symbols with the transmit weights, $W_m(k)$ for $k \in K$, to achieve channel inversion and power distribution. Signal scaling units 1018a through 1018s provide N_C scaled modulation symbol streams.

[1123] Each data stream is transmitted on a respective parallel channel that may include any number and any combination of subbands, transmit antennas, and spatial channels. For example, one data stream may be transmitted on all usable subbands of each wideband eigenmode, as described above. TX spatial processor 928 performs the required spatial processing, if any, on the N_C scaled modulation symbol streams and provides N_{ap} transmit symbol streams. The spatial processing may be performed as shown in Table 1.

[1124] For a transmission scheme whereby one data stream is transmitted on all subbands of each wideband eigenmode (for a full-CSI MIMO system, as described above), N_S sets of encoder 1012, channel interleaver 1014, symbol mapping unit 1016, and signal scaling unit 1018 may be used to process N_S data streams (where $N_C = N_S = N_{ap} \leq N_{ut}$ for a full rank channel response matrix) to provide N_{ap} scaled modulation symbol streams. TX spatial processor 928 then performs spatial processing on the N_{ap} scaled modulation symbol streams, as shown in Table 1, to provide the N_{ap} transmit symbol streams.

[1125] For a transmission scheme whereby one data stream is transmitted on all subbands of each transmit antenna (for a partial-CSI MIMO system), N_{ap} sets of encoder

1012, channel interleaver 1014, symbol mapping unit 1016, and signal scaling unit 1018 may be used to process N_{ap} data streams (where $N_c = N_{ap}$) to provide N_{ap} scaled modulation symbol streams. TX spatial processor 928 then simply passes each scaled modulation symbol stream as a transmit symbol stream. Since spatial processing is not performed for this transmission scheme, each transmit symbol is a modulation symbol.

[1126] In general, TX spatial processor 928 performs the appropriate demultiplexing and/or spatial processing of the scaled modulation symbols to obtain transmit symbols for the parallel channel used for each data stream. TX spatial processor 928 further multiplexes pilot symbols with the transmit symbols, e.g., using time division multiplex (TDM) or code division multiplex (CDM). The pilot symbols may be sent in all or a subset of the subbands/eigenmodes used to transmit traffic data. TX spatial processor 928 provides N_{ap} transmit symbol streams to N_{ap} transmitter units 930a through 930ap.

[1127] Each transmitter unit 930 performs OFDM processing on a respective transmit symbol stream and provides a corresponding modulated signal. The OFDM processing typically includes (1) transforming each set of N_F transmit symbols to the time domain using an N_F -point inverse fast Fourier transform (IFFT) to obtain a "transformed" symbol that contains N_F samples and (2) repeating a portion of each transformed symbol to obtain an OFDM symbol that contains $N_F + N_{cp}$ samples. The repeated portion is referred to as the cyclic prefix, and N_{cp} indicates the number of samples being repeated. The OFDM symbols are further processed (e.g., converted to one or more analog signals, amplified, filtered, and frequency upconverted) by transmitter unit 930 to generate the modulated signal. Other designs for transmitter subsystem 1000 may also be implemented and are within the scope of the invention.

[1128] Controller 940 may perform various functions related to closed-loop rate control for the downlink and uplink (e.g., transmission mode selection for the uplink and transmit weight computation for the downlink). For uplink transmission, controller 940 may perform process 800 in FIG. 8 and selects a transmission mode for each of the multiple parallel channels on the uplink. Within controller 940, a power allocation unit 1042 distributes the total transmit power, $P_{\text{total, up}}$, to the multiple parallel channels (e.g., based on the channel gain estimates $\hat{\sigma}_m(k)$ and the noise floor estimate $N_{0,ap}$ for the access point). A channel inversion unit 1044 performs channel inversion for each

parallel channel. A transmission mode (TM) selector 1046 selects a suitable transmission mode for each parallel channel. Memory unit 942 may store a look-up table 1048 for supported transmission modes and their required SNRs (e.g., as shown in Table 2). For downlink transmission, controller 940 may also perform process 800 in FIG. 8 to determine the transmit power for each subband of each wideband eigenmode and computes the transmit weights used for scaling modulation symbols prior to transmission on the downlink.

[1129] FIG. 11 shows a block diagram of a receiver subsystem 1100, which is an embodiment of the receiver portion of user terminal 520x. The N_{ap} transmitted signals from access point 510x are received by antennas 952a through 952ut, and the received signal from each antenna is provided to a respective receiver unit 954. Each receiver unit 954 conditions and digitizes its received signal to obtain a stream of samples, and further performs OFDM processing on the samples. The OFDM processing at the receiver typically includes (1) removing the cyclic prefix in each received OFDM symbol to obtain a received transformed symbol and (2) transforming each received transformed symbol to the frequency domain using a fast Fourier transform (FFT) to obtain a set of N_F received symbols for the N_F subbands. The received symbols are estimates of the transmit symbols sent by access point 510x. Receiver units 954a through 954ut provide N_{ur} received symbol streams to RX spatial processor 962.

[1130] RX spatial processor 962 performs spatial or space-time processing on the N_{ur} received symbol streams to provide N_C recovered symbol streams. RX spatial processor 962 may implement a linear zero-forcing (ZF) equalizer (which is also referred to as a channel correlation matrix inversion (CCMI) equalizer), a minimum mean square error (MMSE) equalizer, an MMSE linear equalizer (MMSE-LE), a decision feedback equalizer (DFE), or some other equalizer.

[1131] RX data processor 964 receives the N_C recovered symbol streams from RX spatial processor 962. Each recovered symbol stream is provided to a respective symbol demapping unit 1132, which demodulates the recovered symbols in accordance with the modulation scheme used for that stream, as indicated by a demodulation control provided by controller 970. The demodulated data stream from each symbol demapping unit 1132 is de-interleaved by an associated channel de-interleaver 1134 in a manner complementary to that performed at access point 510x for that data stream. If the interleaving is dependent on transmission mode, then controller 970 provides a

deinterleaving control to channel de-interleaver 1134, as indicated by the dashed line. The de-interleaved data from each channel de-interleaver 1134 is decoded by an associated decoder 1136 in a manner complementary to that performed at access point 510x, as indicated by a decoding control provided by controller 970. For example, a turbo decoder or a Viterbi decoder may be used for decoder 1136 if turbo or convolutional coding, respectively, is performed at access point 510x. Decoder 1136 may also provide the status of each received data packet (e.g., indicating whether the packet was received correctly or in error). Decoder 1136 may further store demodulated data for packets decoded in error, so that this data may be combined with additional data from a subsequent incremental transmission and decoded.

[1132] In the embodiment shown in FIG. 11, channel estimator 974 estimates the channel response and the noise floor at user terminal 520x (e.g., based on the received pilot symbols) and provides the channel estimates to controller 970. Controller 970 performs various functions related to closed-loop rate control for both the downlink and uplink (e.g., transmission mode selection for the downlink and transmit weight computation for the uplink). For downlink transmission, controller 970 may perform process 800 in FIG. 8. Within controller 970, a power allocation unit 1172 distributes the total transmit power, $P_{\text{total, dn}}$, to the multiple parallel channels (e.g., based on the channel gain estimates $\hat{\sigma}_m(k)$ and the noise floor $N_{0, \text{ut}}$ estimate for the user terminal). A channel inversion unit 1174 performs channel inversion for each of the multiple parallel channels. A transmission mode (TM) selector 1176 selects a suitable transmission mode for each parallel channel. Memory unit 972 may store a look-up table 1178 for supported transmission modes and their required SNRs (e.g., as shown in Table 2). Controller 970 provides N_C selected transmission modes for the N_C parallel channels on the downlink, which may be part of the feedback information sent to access point 510x. For uplink transmission, controller 970 may also perform process 800 in FIG. 8 to determine the transmit power for each subband of each wideband eigenmode and computes the transmit weights used for scaling modulation symbols prior to transmission on the uplink.

[1133] For clarity, transmitter subsystem 1000 has been described for access point 510x and receiver subsystem 1100 has been described for user terminal 520x. Transmitter subsystem 1000 may also be used for the transmitter portion of user

terminal 520x, and receiver subsystem 1100 may also be used for the receiver portion of access point 510x.

B. Downlink and Uplink Rate Control

[1134] FIG. 12A shows a process for performing closed-loop rate control for the downlink based on the frame structure shown in FIG. 6. A BCH PDU is transmitted in the first segment of each TDD frame (see FIG. 6) and includes the MIMO pilot that can be used by the user terminal to estimate and track the downlink. A steered reference may also be sent in the preamble of an FCH PDU sent to the user terminal. The user terminal estimates the downlink based on the MIMO pilot and/or the steered reference and selects a suitable transmission mode (with the highest supported data rate) for each downlink wideband eigenmode (i.e., each parallel channel). The user terminal then sends these transmission modes as “proposed” transmission modes for the downlink in an RCH PDU sent to the access point.

[1135] The access point receives the proposed transmission modes from the user terminal and schedules data transmission on the downlink in subsequent TDD frame(s). The access point selects the transmission modes for the downlink, which may be the ones received from the user terminal or some other transmission modes (with lower data rates), depending on system loading and other factors. The access point sends assignment information for the user terminal (which includes the transmission modes selected by the access point for downlink transmission) on the FCCH. The access point then transmits data on the FCH to the user terminal using the selected transmission modes. The user terminal receives the assignment information and obtains the transmission modes selected by the access point. The user terminal then processes the downlink transmission in accordance with the selected transmission mode. For the embodiment shown in FIG. 12A, the delay between the channel estimation and transmission mode selection by the user terminal and the use of these transmission modes for downlink transmission is typically one TDD frame, but may be different depending on applications, system configurations, and other factors.

[1136] FIG. 12B shows a process for performing closed-loop rate control for the uplink based on the frame structure shown in FIG. 6. The user terminal transmits a steered reference on the RACH during system access and on the RCH upon being assigned FCH/RCH resources (see FIG. 6). The access point estimates the uplink based on the received steered reference and selects a suitable transmission mode for each

uplink wideband eigenmode. The access point sends assignment information for the user terminal (which includes the transmission modes selected for uplink transmission) on the FCCH. The user terminal transmits data on the RCH to the access point using the selected transmission modes. The access point processes the uplink transmission in accordance with the selected transmission modes.

[1137] The closed-loop rate control techniques described herein may be implemented by various means. For example, these techniques may be implemented in hardware, software, or a combination thereof. For a hardware implementation, the elements used for closed-loop rate control at the transmitter and the receiver (e.g., controllers 940 and 970) may be implemented within one or more application specific integrated circuits (ASICs), digital signal processors (DSPs), digital signal processing devices (DSPDs), programmable logic devices (PLDs), field programmable gate arrays (FPGAs), processors, controllers, micro-controllers, microprocessors, other electronic units designed to perform the functions described herein, or a combination thereof.

[1138] For a software implementation, portions of the closed-loop rate control may be implemented with modules (e.g., procedures, functions, and so on) that perform the functions described herein. The software codes may be stored in a memory unit (e.g., memory unit 942 or 972 in FIGS. 9A and 9B) and executed by a processor (e.g., controller 940 or 970). The memory unit may be implemented within the processor or external to the processor, in which case it can be communicatively coupled to the processor via various means as is known in the art.

[1139] Headings are included herein for reference and to aid in locating certain sections. These headings are not intended to limit the scope of the concepts described therein under, and these concepts may have applicability in other sections throughout the entire specification.

[1140] The previous description of the disclosed embodiments is provided to enable any person skilled in the art to make or use the present invention. Various modifications to these embodiments will be readily apparent to those skilled in the art, and the generic principles defined herein may be applied to other embodiments without departing from the spirit or scope of the invention. Thus, the present invention is not intended to be limited to the embodiments shown herein but is to be accorded the widest scope consistent with the principles and novel features disclosed herein.

WHAT IS CLAIMED IS:

CLAIMS

1. A method of transmitting data on a plurality of parallel channels in a wireless communication system, comprising:

obtaining channel estimates for each of the plurality of parallel channels;

selecting a transmission mode for each of the plurality of parallel channels based on the channel estimates for the parallel channel, wherein the transmission mode for each of the plurality of parallel channels indicates a data rate for the parallel channel; and

sending the transmission mode for each of the plurality of parallel channels to a transmitting entity, wherein a data transmission on each of the plurality of parallel channels is processed at the transmitting entity in accordance with the transmission mode selected for the parallel channel.

2. The method of claim 1, further comprising:

receiving data transmissions on the plurality of parallel channels from the transmitting entity; and

processing the data transmissions in accordance with the transmission mode selected for each of the plurality of parallel channels to recover data sent on the parallel channel.

3. The method of claim 1, wherein the channel estimates for each of the plurality of parallel channels include at least one channel gain estimate and a noise floor estimate for the parallel channel.

4. The method of claim 1, wherein the selecting includes

determining a received signal-to-noise ratio (SNR) for each of the plurality of parallel channels based on the channel estimates for the parallel channel, and wherein the transmission mode for each of the plurality of parallel channels is selected based on the received SNR for the parallel channel.

5. The method of claim 4, wherein the selecting further includes determining an SNR offset for each of the plurality of parallel channels, and wherein the transmission mode for each of the plurality of parallel channels is further selected based on the SNR offset for the parallel channel.

6. The method of claim 5, wherein the selecting further includes determining an operating SNR for each of the plurality of parallel channels based on the received SNR and the SNR offset for the parallel channel, and wherein the transmission mode for each of the plurality of parallel channels is selected based on the operating SNR for the parallel channel.

7. The method of claim 6, wherein the transmission mode for each of the plurality of parallel channels is further selected based on a set of required SNRs for a set of transmission modes supported by the system.

8. The method of claim 1, further comprising:
estimating the quality of the data transmission received on each of the plurality of parallel channels, and wherein the transmission mode for each of the plurality of parallel channels is further selected based on the estimated quality of the data transmission received on the parallel channel.

9. The method of claim 5, further comprising:
adjusting the SNR offset for each of the plurality of parallel channels based on status of data packets received on the parallel channel.

10. The method of claim 5, further comprising:
adjusting the SNR offset for each of the plurality of parallel channels based on at least one decoder metric maintained for the parallel channel.

11. The method of claim 1, further comprising:
detecting for packet errors for each of the plurality of parallel channels; and

adjusting the transmission mode for each of the plurality of parallel channels based on the packet errors for the parallel channel.

12. The method of claim 4, further comprising:

determining an SNR margin for each of the plurality of parallel channels based on the received SNR and a required SNR for the parallel channel; and

adjusting the transmission mode for each of the plurality of parallel channels based on SNR margins for the plurality of parallel channels.

13. The method of claim 6, further comprising:

distributing total transmit power to the plurality of parallel channels, and wherein the operating SNR for each of the plurality of parallel channels is further determined based on transmit power distributed to the parallel channel.

14. The method of claim 13, wherein the total transmit power is uniformly distributed to the plurality of parallel channels.

15. The method of claim 13, wherein the total transmit power is distributed to the plurality of parallel channels using a water-filling procedure.

16. The method of claim 13, further comprising:

determining excess power for each of the plurality of parallel channels based on the operating SNR for the parallel channel, a required SNR for the transmission mode selected for the parallel channel, and the transmit power distributed to the parallel channel;

accumulating the excess power for each of the plurality of parallel channels to obtain total excess power for the plurality of parallel channels; and

redistributing the total excess power to at least one of the plurality of parallel channels.

17. The method of claim 16, wherein the total excess power is redistributed evenly to unsaturated parallel channels among the plurality of parallel channels, where the unsaturated parallel channels have data rates greater than zero and less than a maximum data rate.

18. The method of claim 16, wherein the total excess power is redistributed to one parallel channel, selected from among the plurality of parallel channels, that can achieve a highest increase in data rate with the total excess power.

19. The method of claim 13, wherein each of the plurality of parallel channels includes a plurality of subbands, the method further comprising:

distributing the transmit power for each of the plurality of parallel channels across the plurality of subbands of the parallel channel to achieve similar received SNRs for the plurality of subbands.

20. The method of claim 13, wherein each of the plurality of parallel channels includes a plurality of subbands, the method further comprising:

distributing the transmit power for each of the plurality of parallel channels uniformly across the plurality of subbands of the parallel channel.

21. The method of claim 1, wherein the wireless communication system is an orthogonal frequency division multiplex (OFDM) communication system, and wherein the plurality of parallel channels are formed by a plurality of disjoint sets of subbands.

22. The method of claim 1, wherein the wireless communication system is a frequency division multiplex (FDM) communication system, and wherein the plurality of parallel channels are formed by a plurality of frequency subbands.

23. The method of claim 1, wherein the wireless communication system is a time division multiplex (TDM) communication system, and wherein the plurality of parallel channels are formed by a plurality of time slots.

24. The method of claim 1, wherein the wireless communication system is a multiple-input multiple-output (MIMO) communication system, and wherein the plurality of parallel channels are formed by a plurality of spatial channels.

25. The method of claim 1, wherein the wireless communication system is a multiple-input multiple-output (MIMO) communication system with orthogonal frequency division multiplex (OFDM).

26. The method of claim 25, wherein the plurality of parallel channels are formed by a plurality of wideband spatial channels, and wherein each of the plurality of parallel channels includes a plurality of subbands.

27. The method of claim 25, wherein the channel estimates for each of the plurality of parallel channels are obtained based on a pilot transmitted from each of a plurality of antennas by the transmitting entity.

28. The method of claim 25, wherein the channel estimates for each of the plurality of parallel channels are obtained based on a steered reference transmitted from a plurality of antennas by the transmitting entity.

29. An apparatus in a wireless communication system, comprising:
means for obtaining channel estimates for each of a plurality of parallel channels;

means for selecting a transmission mode for each of the plurality of parallel channels based on the channel estimates for the parallel channel, wherein the transmission mode for each of the plurality of parallel channels indicates a data rate for the parallel channel; and

means for sending the transmission mode for each of the plurality of parallel channels to a transmitting entity, wherein a data transmission on each of the plurality of parallel channels is processed at the transmitting entity in accordance with the transmission mode selected for the parallel channel.

30. The apparatus of claim 29, further comprising:
means for receiving data transmissions on the plurality of parallel channels from the transmitting entity; and

means for processing the received data transmissions in accordance with the transmission mode selected for each of the plurality of parallel channels to recover data sent on the parallel channel.

31. The apparatus of claim 29, wherein the means for selecting includes

means for determining a received signal-to-noise ratio (SNR) for each of the plurality of parallel channels based on the channel estimates for the parallel channel, and wherein the transmission mode for each of the plurality of parallel channels is selected based on the received SNR for the parallel channel.

32. The apparatus of claim 29, further comprising:

means for estimating the quality of the data transmission received on each of the plurality of parallel channels, and wherein the transmission mode for each of the plurality of parallel channels is further selected based on the estimated quality of the data transmission received on the parallel channel.

33. An apparatus in a wireless communication system, comprising:

a channel estimator operative to obtain channel estimates for each of a plurality of parallel channels; and

a controller operative to select a transmission mode for each of the plurality of parallel channels based on the channel estimates for the parallel channel, wherein the transmission mode for each of the plurality of parallel channels indicates a data rate for the parallel channel, and wherein a data transmission on each of the plurality of parallel channels is processed at a transmitting entity in accordance with the transmission mode selected for the parallel channel.

34. The apparatus of claim 33, further comprising:

a receive (RX) data processor operative to receive data transmissions on the plurality of parallel channels and to process the received data transmissions in accordance with the transmission mode selected for each of the plurality of parallel channels to recover data sent on the parallel channel.

35. The apparatus of claim 33, wherein the controller is operative to determine a received signal-to-noise ratio (SNR) for each of the plurality of parallel channels based on the channel estimates for the parallel channel and to select the transmission mode for each parallel channel based on the received SNR for the parallel channel.

36. The apparatus of claim 33, wherein the controller is operative to obtain an estimate of the quality of the data transmission received on each of the plurality of parallel channels and to adjust the transmission mode for each parallel channel based on the estimated quality of the data transmission received on the parallel channel.

37. A method of transmitting data on a plurality of parallel channels in a wireless communication system, comprising:

receiving feedback information from a receiving entity, wherein the feedback information is indicative of the quality of the plurality of parallel channels;

determining a transmission mode for each of the plurality of parallel channels based on the feedback information, wherein the transmission mode for each of the plurality of parallel channels indicates a data rate for the parallel channel;

processing data for each of the plurality of parallel channels in accordance with the transmission mode for the parallel channel; and

transmitting the processed data for each of the plurality of parallel channels on the parallel channel to the receiving entity.

38. The method of claim 37, wherein the transmission mode for each of the plurality of parallel channels is selected by the receiving entity based on channel estimates obtained for the parallel channel, and wherein the feedback information includes a plurality of transmission modes selected by the receiving entity for the plurality of parallel channels.

39. The method of claim 37, further comprising:

obtaining channel gain estimates for each of the plurality of parallel channels, and wherein the transmission mode for each of the plurality of parallel channels is determined based on the channel gain estimates for the parallel channel and a noise floor estimate for the parallel channel included in the feedback information from the receiving entity.

40. The method of claim 39, wherein the channel gain estimates for each of the plurality of parallel channels are obtained based on a steered reference received from the receiving entity.

41. The method of claim 37, further comprising:

receiving an adjustment to the transmission mode for a first parallel channel among the plurality of parallel channels; and

processing data for the first parallel channel in accordance with the adjustment to the transmission mode for the first parallel channel.

42. The method of claim 41, wherein the adjustment to the transmission mode for the first parallel channel is determined based on packet errors detected for the first parallel channel.

43. The method of claim 41, wherein the adjustment to the transmission mode for the first parallel channel is determined based on a received signal-to-noise ratio (SNR) and a required SNR for the first parallel channel.

44. The method of claim 37, further comprising:

computing, for each of the plurality of parallel channels, a plurality of transmit weights for a plurality of subbands of the parallel channel, wherein the plurality of transmit weights achieve similar received signal-to-noise ratios (SNRs) for the plurality of subbands of the parallel channel; and

scaling the processed data for each of the plurality of parallel channels with the plurality of transmit weights for the parallel channel, and wherein the scaled and processed data for each of the plurality of parallel channels is transmitted on the parallel channel.

45. An apparatus in a wireless communication system, comprising:

means for receiving feedback information from a receiving entity, wherein the feedback information is indicative of the quality of the plurality of parallel channels;

means for determining a transmission mode for each of a plurality of parallel channels based on the feedback information, wherein the transmission mode for each of the plurality of parallel channels indicates a data rate for the parallel channel;

means for processing data for each of the plurality of parallel channels in accordance with the transmission mode for the parallel channel; and

means for transmitting the processed data for each of the plurality of parallel channels on the parallel channel.

46. The apparatus of claim 45, further comprising:

means for obtaining channel gain estimates for each of the plurality of parallel channels, and wherein the transmission mode for each of the plurality of parallel channels is determined based on the channel gain estimates for the parallel channel and a noise floor estimate for the parallel channel included in the feedback information from the receiving entity.

47. The apparatus of claim 45, further comprising:

means for receiving an adjustment to the transmission mode for a first parallel channel among the plurality of parallel channels; and

means for processing data for the first parallel channel in accordance with the adjustment to the transmission mode for the first parallel channel

48. An apparatus in a wireless communication system, comprising:

a controller operative to determine a transmission mode for each of a plurality of parallel channels based on feedback information received from a receiving entity, wherein the feedback information is indicative of the quality of the plurality of parallel channels, and wherein the transmission mode for each of the plurality of parallel channels indicates a data rate for the parallel channel;

a transmit (TX) data processor operative to process data for each of the plurality of parallel channels in accordance with the transmission mode for the parallel channel; and

at least one transmitter unit operative to transmit the processed data for each of the plurality of parallel channels on the parallel channel.

49. The apparatus of claim 48, wherein the controller is operative to obtain channel gain estimates for each of the plurality of parallel channels and to determine the transmission mode for each of the plurality of parallel channels based on the channel gain estimates for the parallel channel and a noise floor estimate for the parallel channel included in the feedback information from the receiving entity.

50. The apparatus of claim 48, wherein the controller is operative to obtain an adjustment to the transmission mode for a first parallel channel among the plurality of parallel channels, and wherein the TX data processor is operative to process data for the first parallel channel in accordance with the adjustment to the transmission mode for the first parallel channel.

51. A method of transmitting data on a plurality of parallel channels in a wireless communication system, comprising:

- obtaining channel estimates for each of the plurality of parallel channels;

- computing a received signal-to-noise ratio (SNR) for each of the plurality of parallel channels based on the channel estimates for the parallel channel;

- computing an operating SNR for each of the plurality of parallel channels based on the received SNR and an SNR offset for the parallel channel;

- selecting a transmission mode for each of the plurality of parallel channels based on the operating SNR for the parallel channel and a set of required SNRs for a set of transmission modes supported by the system, wherein the transmission mode for each of the plurality of parallel channels indicates a data rate for the parallel channel; and

- processing data for each of the plurality of parallel channels in accordance with the transmission mode selected for the parallel channel.

52. The method of claim 51, further comprising:

estimating the quality of a data transmission received on each of the plurality of parallel channels; and

adjusting the SNR offset for each of the plurality of parallel channels based on the estimated quality of the data transmission received on the parallel channel.

53. The method of claim 52, wherein the quality of the data transmission received on each of the plurality of parallel channels is estimated based on status of packets received on the parallel channel.

54. The method of claim 52, further comprising:

adjusting the transmission mode for each of the plurality of parallel channels based on the estimated quality of the data transmission received on the parallel channel.

55. An apparatus in a wireless communication system, comprising:

means for obtaining channel estimates for each of a plurality of parallel channels;

means for computing a received signal-to-noise ratio (SNR) for each of the plurality of parallel channels based on the channel estimates for the parallel channel;

means for computing an operating SNR for each of the plurality of parallel channels based on the received SNR and an SNR offset for the parallel channel;

means for selecting a transmission mode for each of the plurality of parallel channels based on the operating SNR for the parallel channel and a set of required SNRs for a set of transmission modes supported by the system, wherein the transmission mode for each of the plurality of parallel channels indicates a data rate for the parallel channel; and

means for processing data for each of the plurality of parallel channels in accordance with the transmission mode selected for the parallel channel.

56. The apparatus of claim 55, further comprising:

means for estimating the quality of a data transmission received on each of the plurality of parallel channels; and

means for adjusting the SNR offset for each of the plurality of parallel channels based on the estimated quality of the data transmission received on the parallel channel.

57. The method of claim 56, further comprising:

means for adjusting the transmission mode for each of the plurality of parallel channels based on the estimated quality of the data transmission received on the parallel channel.

58. An apparatus in a wireless communication system, comprising:

a channel estimator operative to provide channel gain estimates for each of a plurality of parallel channels;

a selector operative to compute a received signal-to-noise ratio (SNR) for each of the plurality of parallel channels based on the channel estimates for the parallel channel, compute an operating SNR for each of the plurality of parallel channels based on the received SNR and an SNR offset for the parallel channel, and select a transmission mode for each of the plurality of parallel channels based on the operating SNR for the parallel channel and a set of required SNRs for a set of transmission modes supported by the system, wherein the transmission mode for each of the plurality of parallel channels indicates a data rate for the parallel channel; and

a data processor operative to process data for each of the plurality of parallel channels in accordance with the transmission mode selected for the parallel channel.

59. The apparatus of claim 58, wherein the selector is operative to receive an estimate of the quality of a data transmission received on each of the plurality of parallel channels and to adjust the SNR offset for each of the plurality of parallel channels based on the estimated quality of the data transmission received on the parallel channel.

60. The method of claim 59, wherein the selector is further operative to adjust the transmission mode for each of the plurality of parallel channels based on the estimated quality of the data transmission received on the parallel channel.

61. A processor readable media for storing instructions operable to:

- obtain channel gain estimates for each of a plurality of parallel channels in a wireless communication system;
- compute a received signal-to-noise ratio (SNR) for each of the plurality of parallel channels based on the channel estimates for the parallel channel;
- compute an operating SNR for each of the plurality of parallel channels based on the received SNR and an SNR offset for the parallel channel; and
- select a transmission mode for each of the plurality of parallel channels based on the operating SNR for the parallel channel and a set of required SNRs for a set of transmission modes supported by the system, wherein the transmission mode for each of the plurality of parallel channels indicates a data rate for the parallel channel, and wherein data is sent on each of the plurality of parallel channels in accordance with the transmission mode selected for the parallel channel.

62. The processor readable media of claim 61 and further storing instructions operable to:

- adjust the SNR offset for each of the plurality of parallel channels based on an estimate of the quality of the data transmission received on the parallel channel.

63. The processor readable media of claim 62 and further storing instructions operable to:

- adjust the transmission mode for each of the plurality of parallel channels based on the estimated quality of the data transmission received on the parallel channel.

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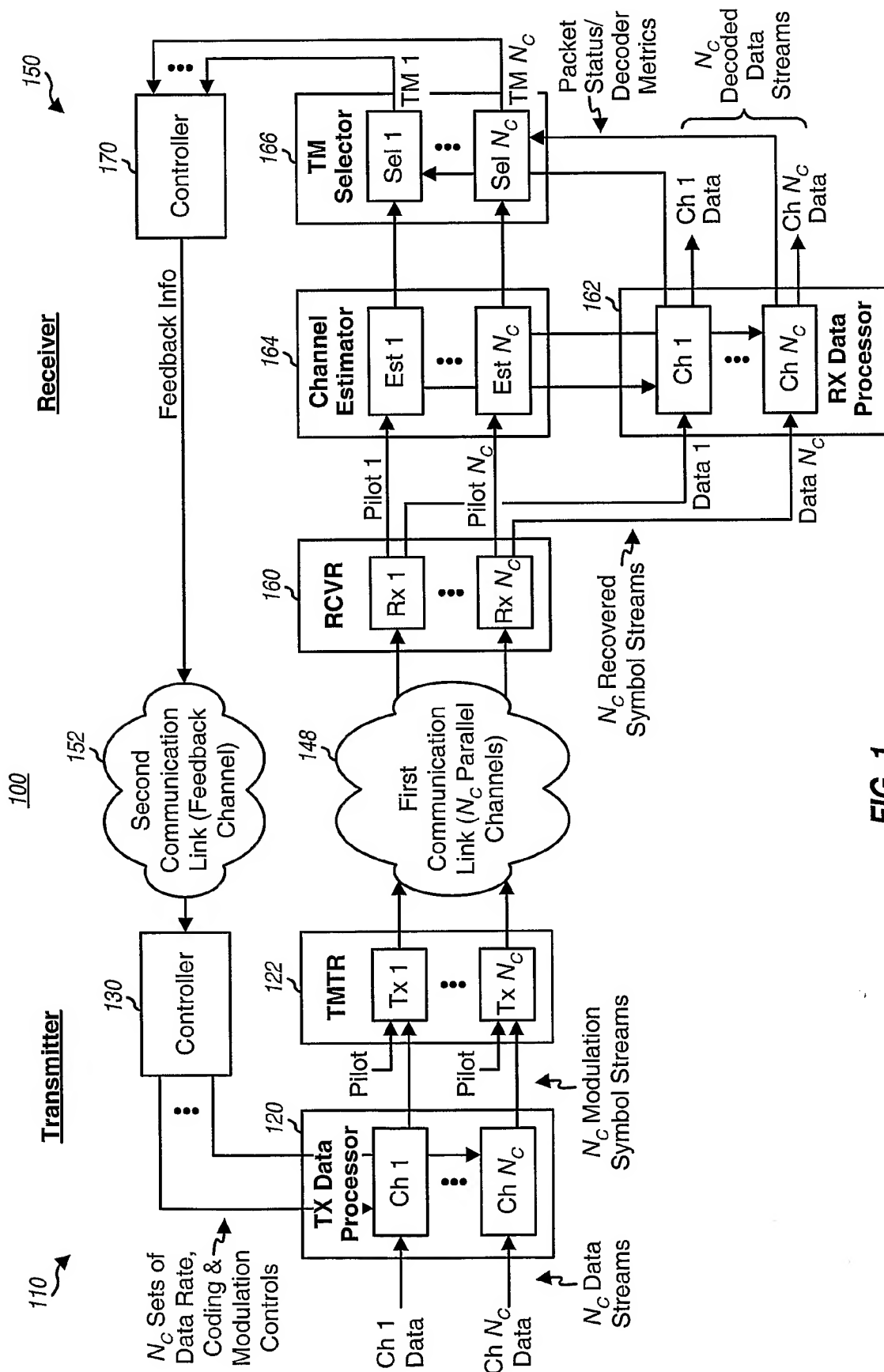
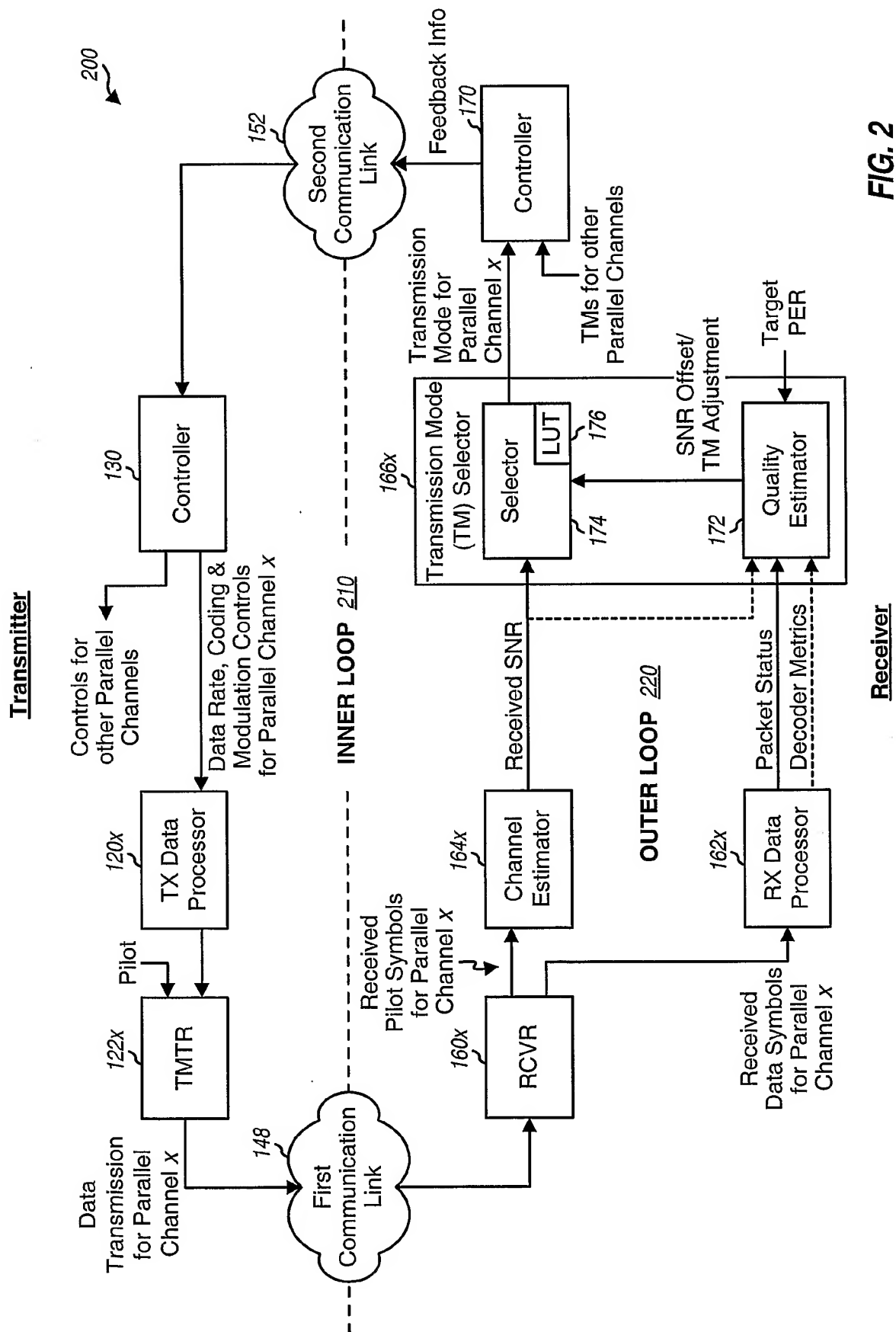


FIG. 1

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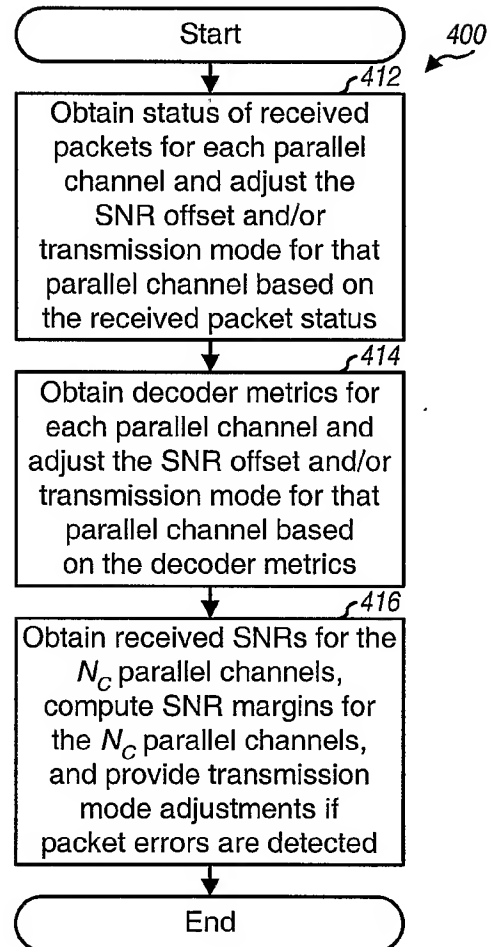
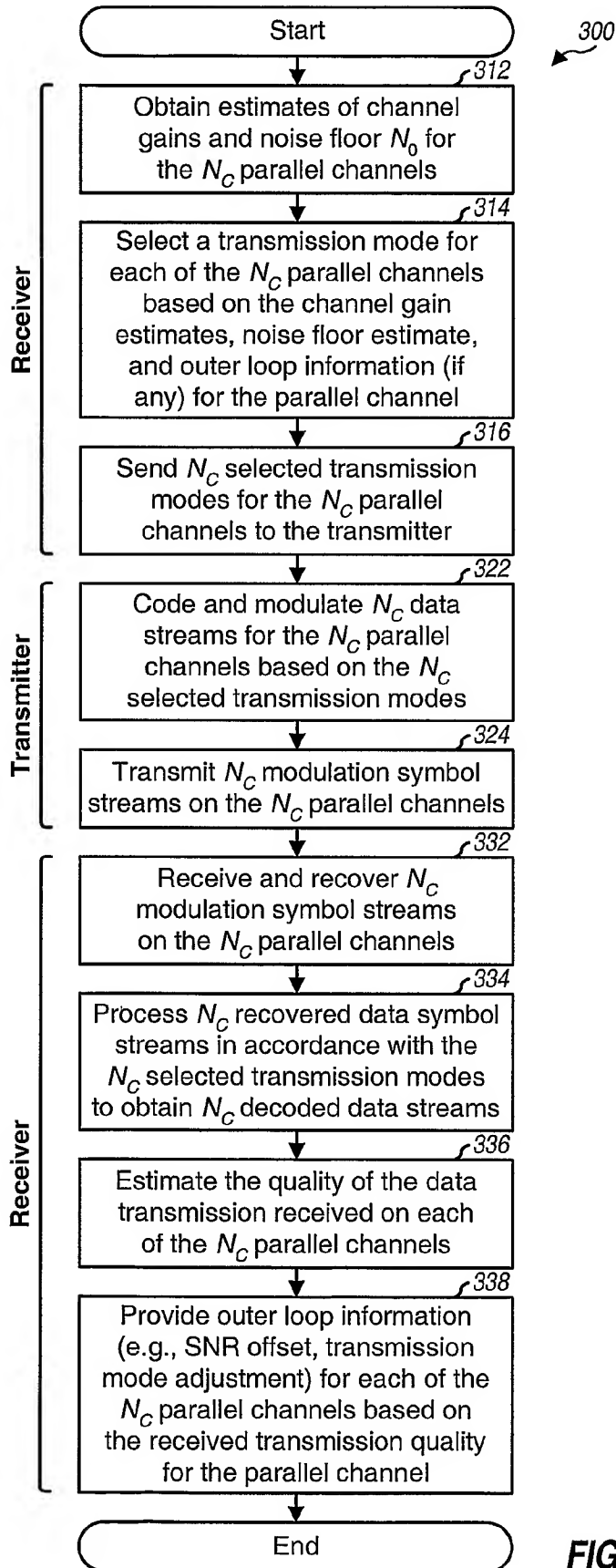


FIG. 4

FIG. 3

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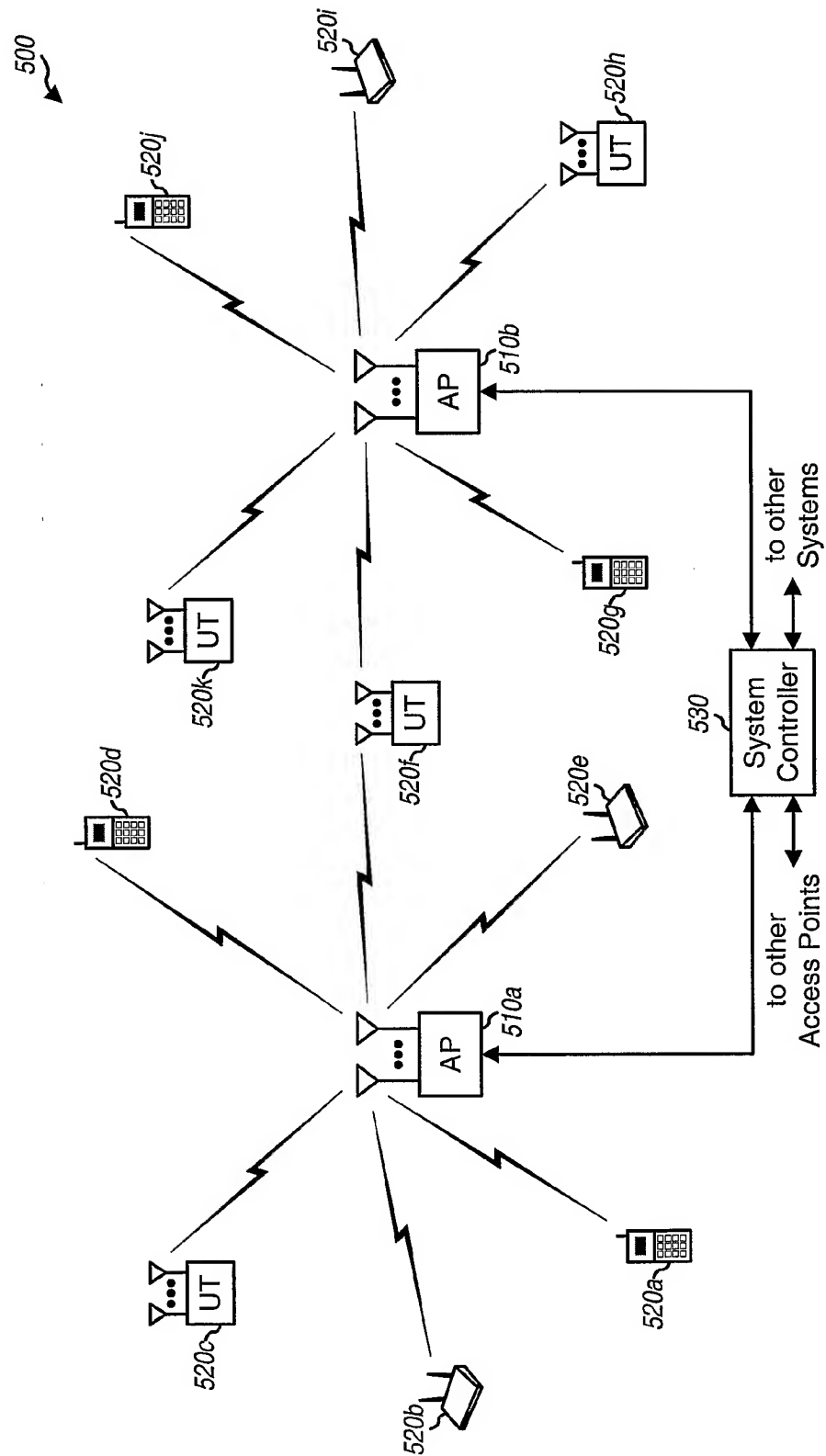


FIG. 5

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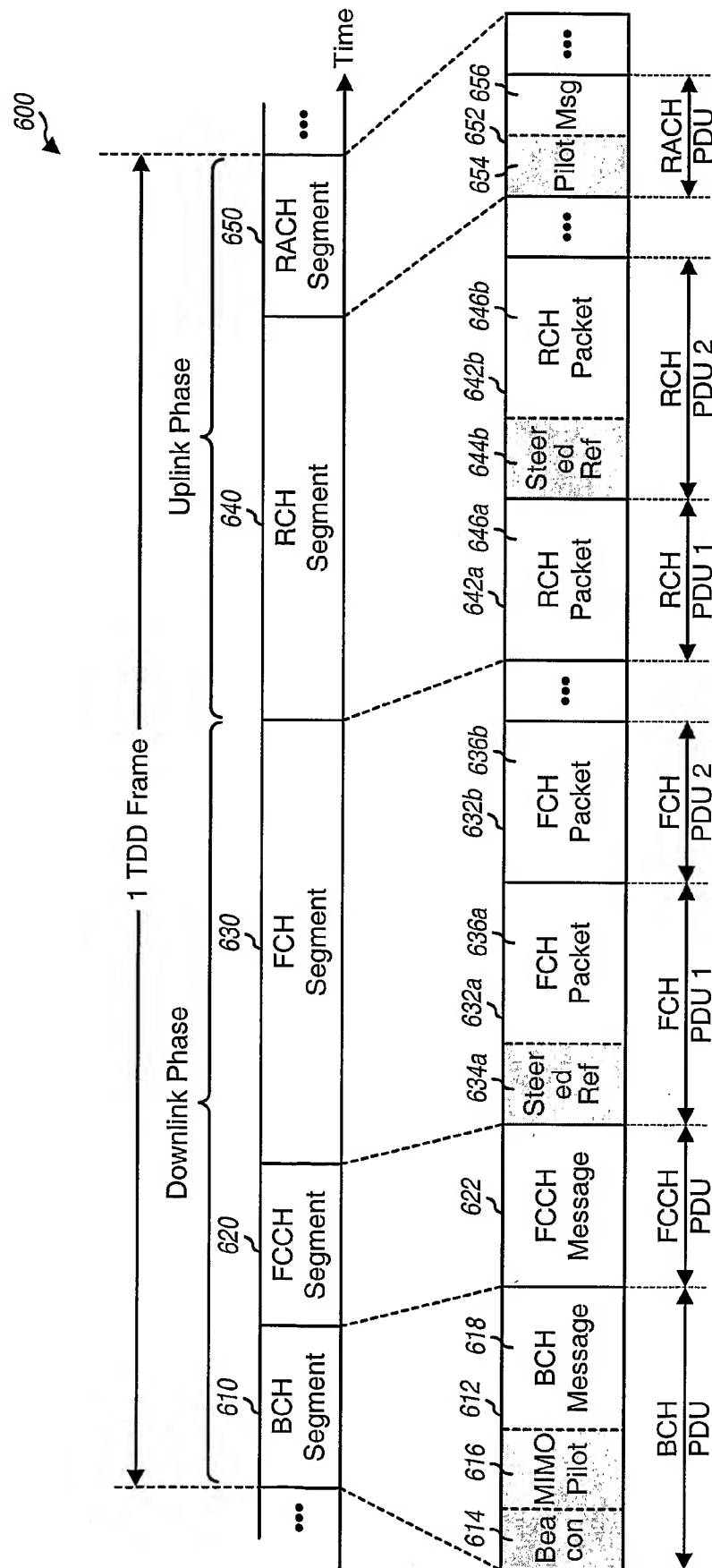
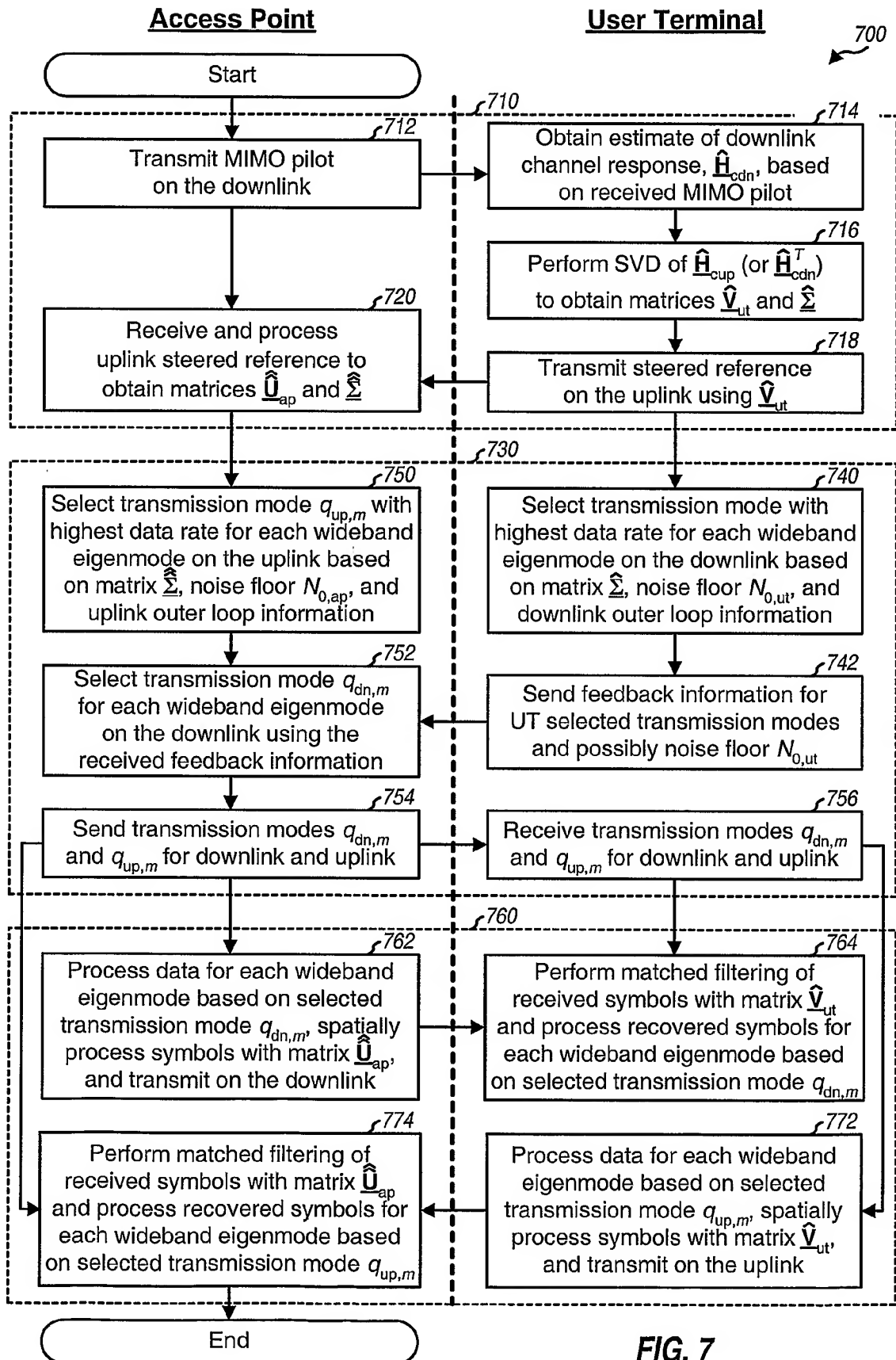


FIG. 6

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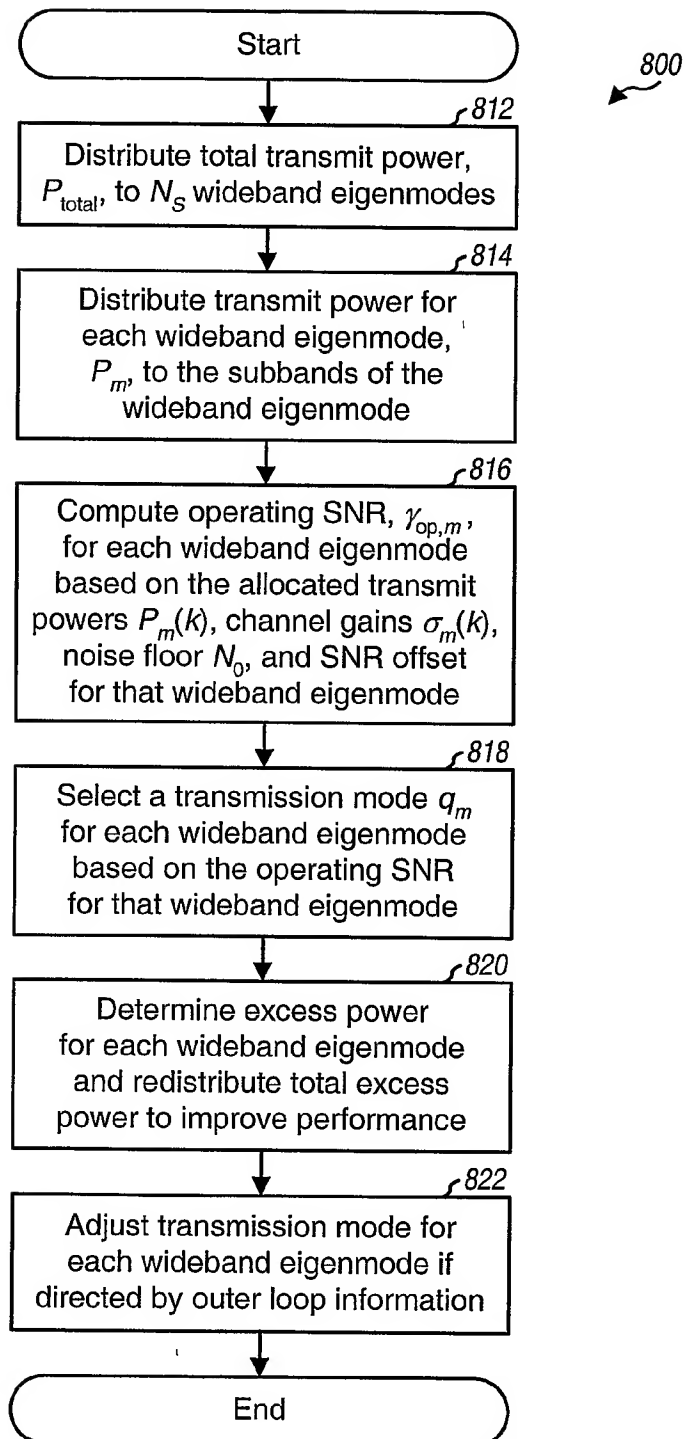


FIG. 8

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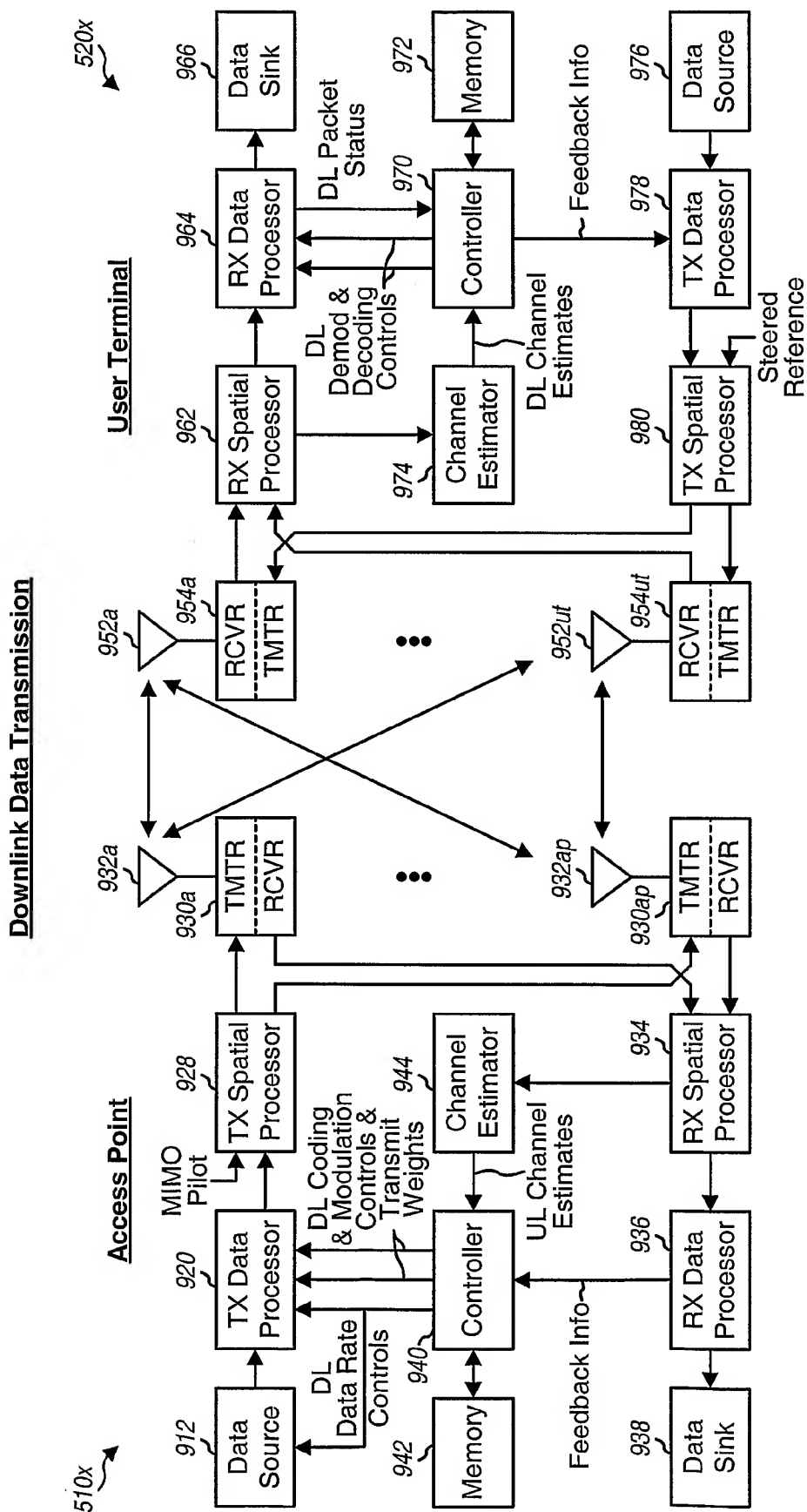


FIG. 9A

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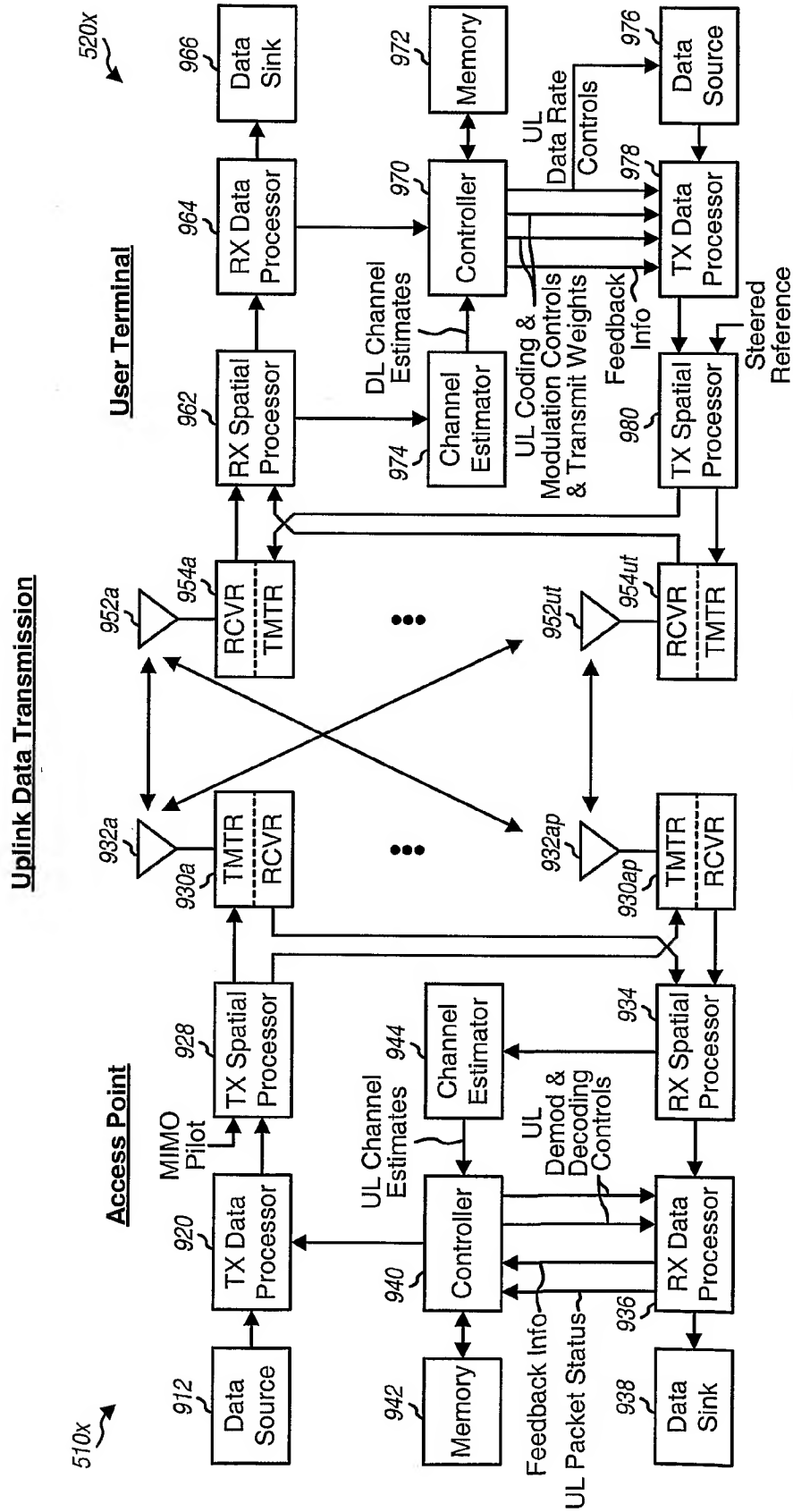


FIG. 9B

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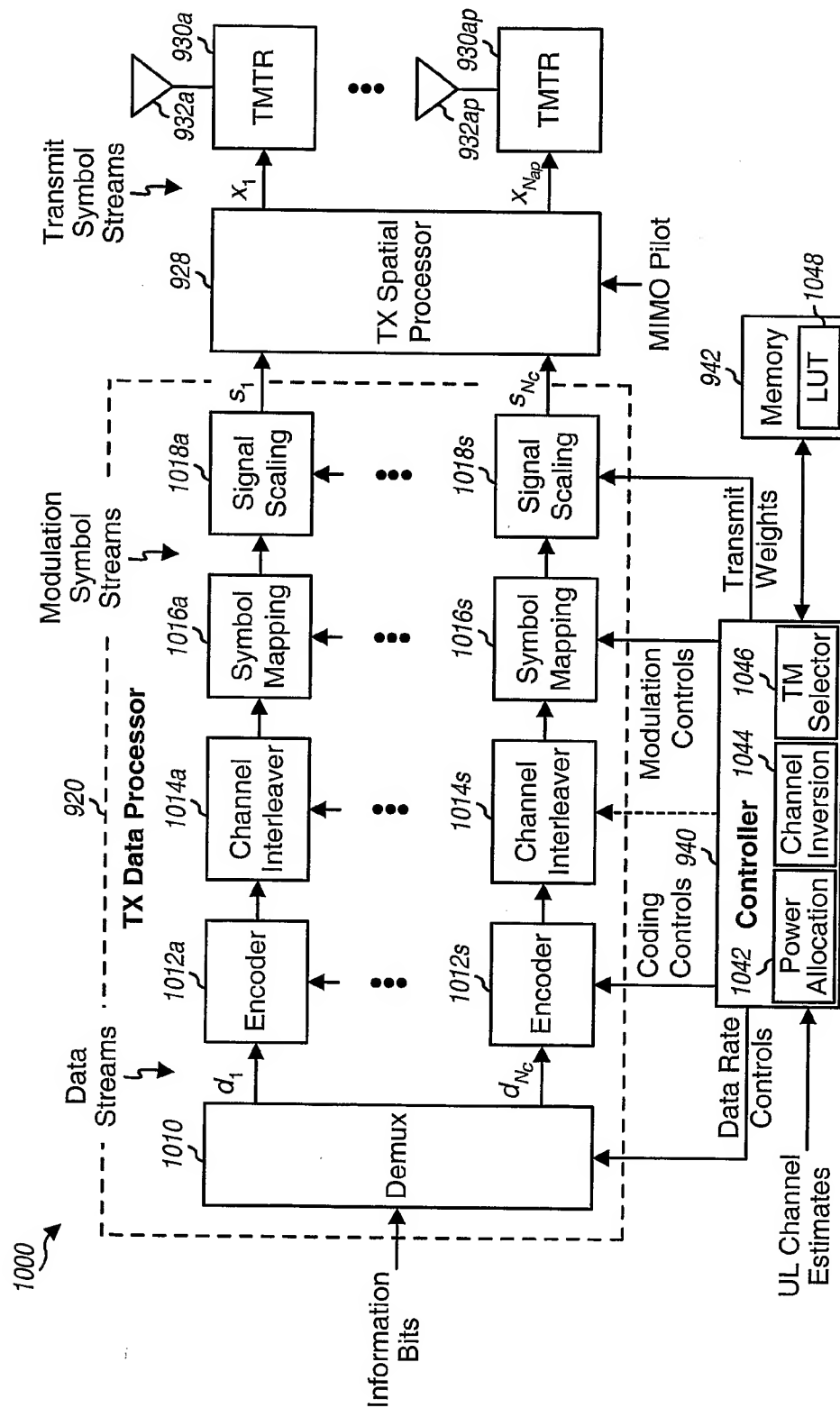


FIG. 10

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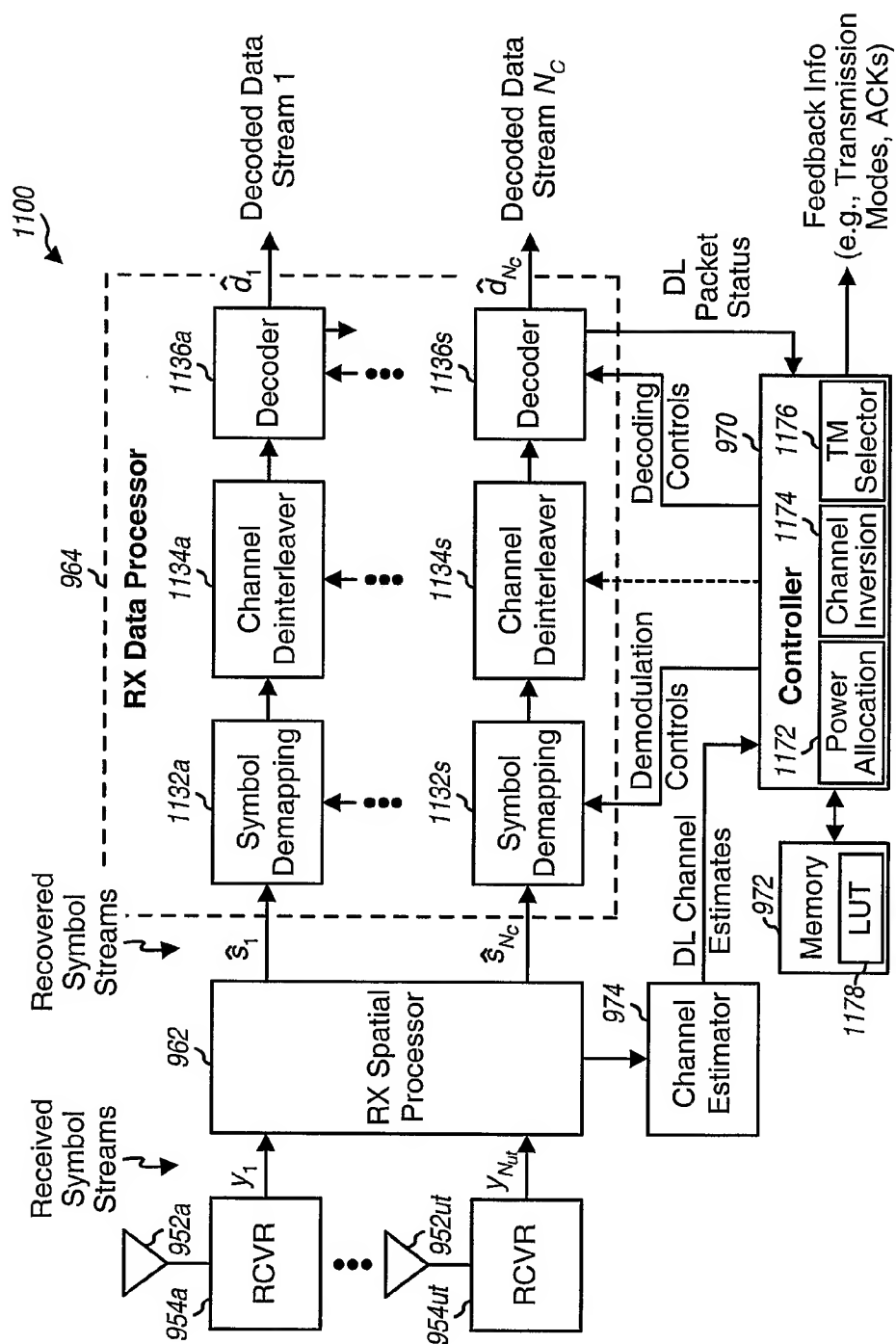
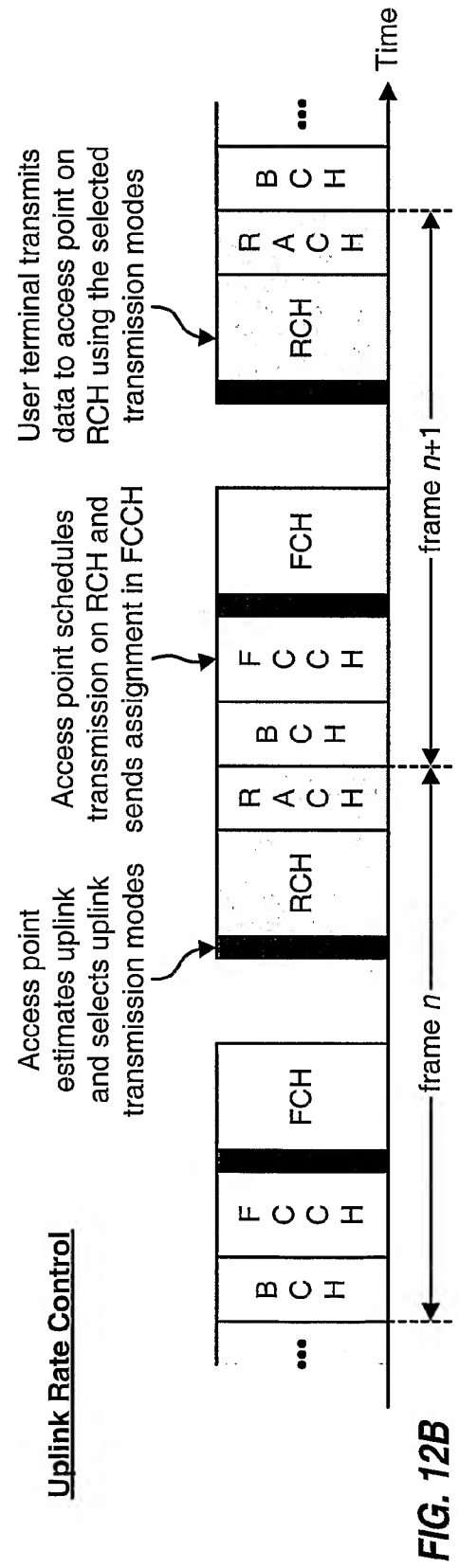
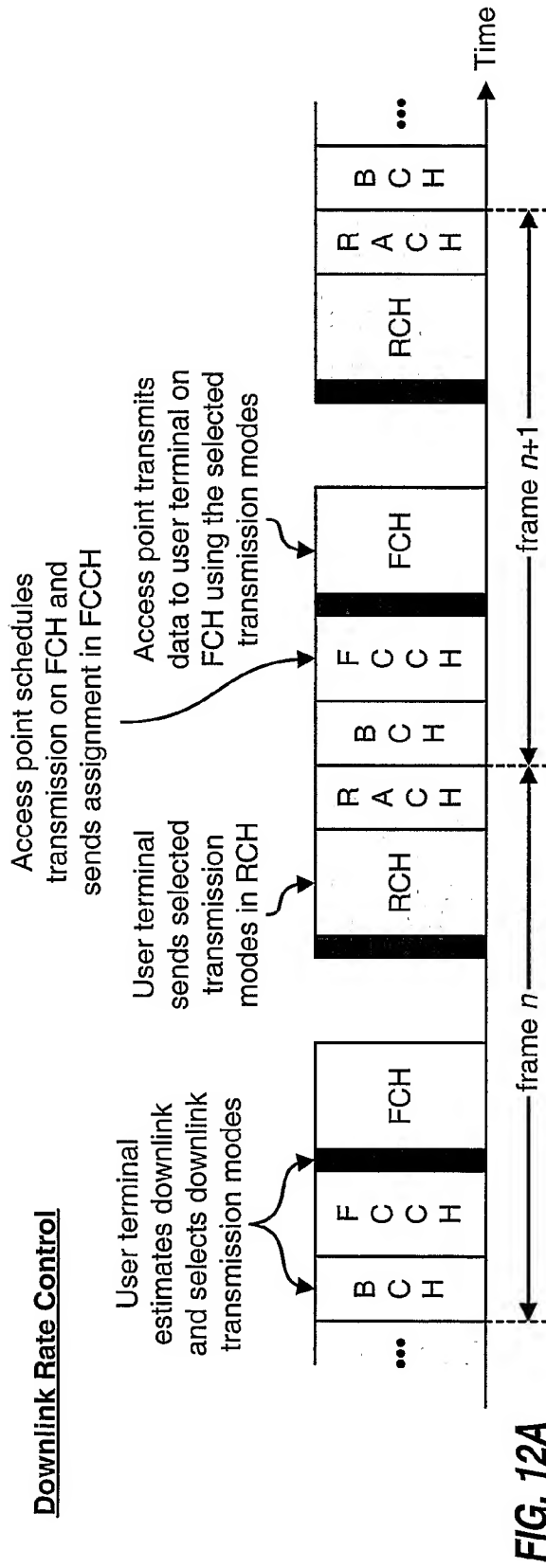


FIG. 11

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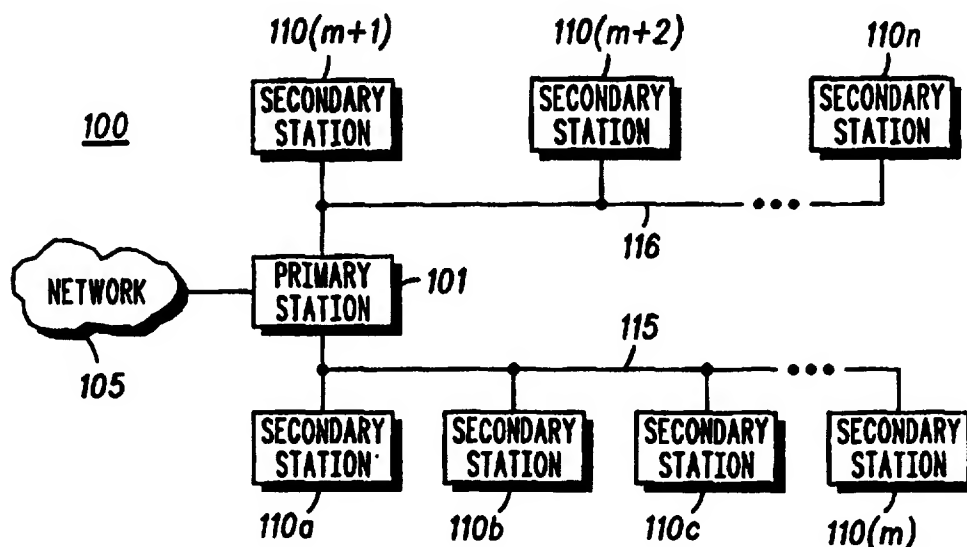




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(54) Title: APPARATUS AND METHOD FOR ADAPTIVE FORWARD ERROR CORRECTION IN DATA COMMUNICATIONS



(57) Abstract

An apparatus (101, 110) and method for adaptive forward error correction in a data communication system (100) provides for dynamically changing forward error correction parameters based upon communication channel conditions. Data having a current degree of forward error correction is received (305), and a channel parameter is monitored (310). A threshold level for the channel parameter is determined (315), and the monitored channel parameter is compared to the threshold level (320). When the channel parameter is not within a predetermined or adaptive variance of the threshold level, a revised forward error correction parameter having a greater or lesser degree of forward error correction capability is selected (330, 340, 350, 360), and the revised forward error correction parameter is transmitted (370). The device receiving the revised forward error correction parameter, such as a secondary station (110), then transmits data encoded utilizing the revised error correction parameter (425).

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APPARATUS AND METHOD FOR ADAPTIVE FORWARD ERROR CORRECTION IN DATA COMMUNICATIONS

Field of the Invention

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This invention relates, in general, to data communications and data communications systems and devices and, more specifically, to an apparatus and method for adaptive forward error correction in data communications.

10

Background of the Invention

With the advent of multimedia communications, data transmission has become increasingly complex. For example, multimedia communications applications such as real time transmission of digitally encoded video, voice, and other forms of data, may require new forms and systems for data communication and data transmission. One such new communication system is the CableComm™ System currently being developed by Motorola, Inc. In the CableComm™ System, a hybrid optical fiber and coaxial cable is utilized to provide substantial bandwidth over existing cable lines to secondary stations such as individual, subscriber access units, for example, households having new or preexisting cable television capability. These coaxial cables are further connected to fiber optical cables to a central location having centralized, primary (or "head end") controllers or stations having receiving and transmitting capability. Such primary equipment may be connected to any variety of networks or other information sources, from the Internet, various on line services, telephone networks, to video/movie subscriber service. With the CableComm™ System, digital data may be transmitted both in the downstream direction, from the primary station or controller (connected to a network) to the secondary station of an individual user (subscriber access unit), and in the upstream

direction, from the secondary station to the primary station (and to a network).

In the CableComm™ System, downstream data is currently intended to be transmitted using 64 quadrature amplitude modulation ("QAM") at a rate of 30 M bps (megabits per second), over channels having 6 MHz bandwidth in the frequency spectrum of 50 - 750 MHz. Anticipating asymmetrical requirements with large amounts of data tending to be transmitted in the downstream direction rather than the upstream direction, less capacity is provided for upstream data transmission, using $\pi/4$ differential quadrature phase shift keying ($\pi/4$ -DQPSK) modulation in the frequency band from 5 - 42 MHz with a symbol rate of 384 k symbols/sec with 2 bits/symbol. In addition, the communication system is designed to have a multipoint configuration, i.e., many end user secondary stations (also referred to as subscriber access units) transmitting upstream to a primary station, with one or more primary stations transmitting downstream to the secondary stations. The communication system is also designed for asynchronous transmission, with users transmitting and receiving packets of encoded data, such as video or text files. In addition, it is also highly likely that transmission may be bursty, with various users receiving or transmitting data at indeterminate intervals over selected channels in response to polling, contention, or other protocols from the primary station, rather than transmitting a more continuous and synchronous stream of information over a dedicated or circuit switched connection.

For such asynchronous data transmission, it is highly desirable to organize data into recognizable formats or packets for reliable detection by the receivers of the primary station or the secondary station. In the CableComm™ System, the initial portion (or preamble) of the data packet contains timing or synchronization information for accurate data transmission. Following the timing information is encoded

data, which may be encoded for both security (encryption) and for error correction. Following the encoded data are error correction information (as encoded bits) and also additional error detection information in the form of cyclic redundancy check (CRC) bits. One difficulty with inclusion of such error correction information is that such inclusion increases the overall packet size, adding overhead for data transmission and correspondingly decreasing data throughput. Secondly, the inclusion of such error correction information typically increases the system response time or latency, due to the time which may be consumed in the error correction encoding and decoding processes. In addition, there may be situations, such as low noise conditions, in which inclusion of such error correction information may be unnecessary, and higher data throughput may be achieved without the additional overhead of error correction information. Various prior art methods for providing error correction capability, however, typically provided only for a fixed error correction capability, without regard for other opportunities to increase data throughput, for low noise conditions, or for needs to decrease response latency. Accordingly, a need has remained for an apparatus and method to provide for adaptive and flexible error correction capability, providing sufficient error correction for accurate data reception while simultaneously providing for overhead minimization for increased data throughput, and for such an apparatus and method to respond and adapt to potentially changing and variable communication channel conditions.

Brief Description of the Drawings

FIG. 1 is a block diagram illustrating a communication system in accordance with the present invention.

FIG. 2 is a block diagram illustrating a primary station apparatus in accordance with the present invention.

FIG. 3 is a block diagram illustrating a secondary station apparatus in accordance with the present invention.

FIG. 4 is a flow chart illustrating channel monitoring and forward error correction parameter adjustment in accordance with the present invention.

FIG. 5 is a flow chart illustrating forward error correction adjustment and data transmission in accordance with the present invention.

Detailed Description of the Invention

As mentioned above, a need has remained for an apparatus and method to provide for adaptive and flexible error correction capability. The apparatus and method in accordance with the present invention provides such adaptive and flexible error correction capability, providing sufficient error correction for accurate data reception, and also providing for overhead minimization for increased data throughput. The apparatus and method of the present invention is also able to respond and adapt to potentially changing and variable communications channel conditions, such as changes in noise conditions and error rates.

FIG. 1 is a block diagram illustrating a communication system 100 in accordance with the present invention. As illustrated in FIG. 1, a primary station 101, also referred to as a primary transceiver or a primary device, is coupled to a plurality of secondary stations 110_a through 110_n, via communications (or communication) media 115 and 116. In the preferred embodiment, communications media 115 and 116 are hybrid optical fiber and coaxial cable. In other embodiments, the communications media, such as communications media 115 and 116, may be coaxial cable, fiber optic cable, twisted pair wires, and so on, and may also include air, atmosphere or space for wireless and satellite communication. The primary station 101 is also coupled to a network 105, which may include

networks such as the Internet, on line services, telephone and cable networks, and other communication systems. The secondary stations 110_a through 110_n are illustrated in FIG. 1 as connected to the primary station 101 on two segments or branches of a communications medium, such as communications media 115 and 116. Equivalently, the secondary stations 110_a through 110_n may be connected to more than one primary station, and may be connected to a primary station (such as primary station 101) utilizing more or fewer branches, segments or sections of any communications medium.

Continuing to refer to FIG. 1, in the preferred embodiment, the communications medium, such as communications media 115 and 116, has or supports a plurality of communications channels. For ease of reference, the communications channels in which a primary station, such as the primary station 101, transmits information, signals, or other data to a secondary station, such as secondary station 110_n, are referred to as downstream channels or downstream communication channels. Also for ease of reference, the communications channels in which a secondary station, such as secondary station 110_n, transmits information, signals, or other data to a primary station, such as primary station 101, are referred to as upstream channels or upstream communication channels. These various upstream and downstream channels may, of course, be the same physical channel or may be separate physical channels, for example, through time division multiplexing or frequency division multiplexing. These various channels may also be logically divided in other ways, in addition to upstream and downstream directions. As mentioned above, in the preferred embodiment of the CableComm™ System, the communications medium is hybrid fiber coaxial cable, with downstream channels in the frequency spectrum (or band) of 50 - 750 MHz, and with upstream channels in the frequency spectrum of 5 - 42 MHz.

FIG. 2 is a block diagram illustrating a primary station 101 in accordance with the present invention. The primary station 101 is coupled to a communication medium 114 for upstream and downstream communication to one or more secondary stations (not illustrated), and is coupleable to a network, such as the Internet, through a network interface 119. The primary station includes a processor arrangement 120 which is connected to a plurality of channel interfaces, channel interface 125_a through channel interface 125_n, for communication over the communication medium 114. The processor arrangement 120 includes a master controller 121 having or connected to memory 122, and one or more additional processors 130_{a1} through 130_{n2} and corresponding associated memories 131_{a1} through 131_{n2}. In the preferred embodiment, the master controller 121 is a Motorola M68040 processor, and the memory 122 is 16 MB RAM. The master controller 121 performs a variety of higher level functions in the preferred embodiment, such as spectrum management, routing functions, management of secondary stations, and communication protocol management (such as SNMP management). The master controller 121 is connected to a plurality of other processors, collectively referred to as processors 130 and separately illustrated as processor 130_{a1}, processor 130_{a2}, through processor 130_{n1} and processor 130_{n2}. Each of these processors, processor 130_{a1}, processor 130_{a2}, through processor 130_{n1} and processor 130_{n2}, is also coupled to or contains corresponding memory circuits, memory 131_{a1}, memory 131_{a2}, through memory 131_{n1} and memory 131_{n2}. In the preferred embodiment, each of these processors 130 are also Motorola M68040 processors, while the corresponding memory circuits, memory 131_{a1} through memory 131_{n2}, are 4 MB RAM. In the preferred embodiment, the processors 130 perform such functions related to upstream and downstream data protocols, such as sending a poll message or an acknowledgment message downstream. Each of these

processors 130_{a1} through 130_{n2} of the processor arrangement 120 are connected to corresponding receivers and transmitters of the channel interfaces, channel interface 125_a through channel interface 125_n (collectively referred to as channel interfaces 125), namely, receiver 135_a through receiver 135_n (collectively referred to as receivers 135) and transmitter 136_a through transmitter 136_n (collectively referred to as transmitters 136). In the preferred embodiment, depending upon the functions implemented, each of the receivers 135_a through 135_n may include a Motorola M68302 processor, a Motorola 56000 series digital signal processor, a ZIF SYN integrated circuit, and an LSI Logic L64714 (Reed-Solomon decoder), for demodulation and for decoding forward error correction and cyclic redundancy checks. In the preferred embodiment, also depending upon the functions implemented, each of the transmitters 136_a through 136_n may include a Motorola M68302 processor, a Motorola 56000 series digital signal processor, a ZIF SYN integrated circuit, and an LSI Logic L64711 (Reed-Solomon encoder), for modulation and for coding for forward error correction and cyclic redundancy checks. As a consequence, as used herein, the channel interfaces 125 may be considered to perform the functions of data and other signal reception and transmission, regardless of the specific hardware implementations and additional functions which may or may not be implemented. The various memories illustrated, such as memory 122 or 131_{a1}, may also be embodied or contained within their corresponding processors, such as master controller 121 or processor 130_{a1}. The functions of these various components with respect to the present invention are explained in greater detail below with reference to FIGs. 4 and 5.

FIG. 3 is a block diagram illustrating a representative secondary station 110_n (of the plurality of secondary stations 110) in accordance with the present invention. The secondary station 110_n includes a processor (or processor arrangement)

150, with the processor 150 having or coupled to a memory 155. In the preferred embodiment, the processor 150 is a Motorola M68302 processor (also known as an integrated multiprotocol processor), and the memory 155 is 256 K RAM.

5 The processor 150 is coupled to an interface 170, such as an ethernet port or an RS232 interface, for connection to a computer, a workstation, or other data terminal equipment ("DTE"). The processor 150 is also coupled to a channel interface 160 for communication over the communication

10 medium 114. The channel interface 160, in the preferred embodiment, depending upon the functions implemented, includes a Motorola M68HC11 integrated circuit, a ZIF SYN integrated circuit, a Broadcom BCM3100 QAMLink integrated circuit, a Motorola TxMod integrated circuit, and LSI Logic

15 L64711 and L64714 integrated circuits, and performs such functions as forward error correction encoding and decoding, QAM demodulation (for downstream reception), QPSK modulation (for upstream transmission), transmit level and frequency adjustment, for data and other signal reception and

20 transmission. As a consequence, as used herein, the channel interface 160 may be considered to perform the functions of data and other signal reception and transmission, regardless of the specific hardware implementations and additional functions which may or may not be implemented. The memory

25 illustrated as memory 155 may also be embodied or contained within the corresponding processor 150. The additional functions of these components of a secondary station 110_n with respect to the invention are also described in greater detail below with reference to FIGs. 4 and 5.

30 As mentioned above, the upstream channels of the communication medium, in the preferred CableCommTM System, are in a frequency range between 5 and 42 MHz and may be susceptible to interference from typical noise sources. Forward error correction is preferably employed on the

35 upstream channels as a way of compensating for data

transmission errors which may have been caused by noise or other distortions. Forward error correction comprises an error correcting code that is added to the user data to allow a receiver to correct certain types and sizes of errors that may have occurred during the transmission of the data. The transmitting unit, such as the processor 150 and channel interface 160 of a secondary station 110_n, generates the error correcting code from the user data, and appends the encoded data onto the user data during transmission. The receiving unit, such as receiver 135_n and processor 130_{n2} of the primary station 101, uses the encoded data to detect received errors and to correct detected errors. As a consequence, the receiving unit should know, prior to the receipt of actual data, what type of error correcting code is to be employed by the transmitting unit, for proper decoding and error correction. This may be typically done by prior agreement (e.g., during initial set up or configuration of the communication system), or during a negotiation "handshake" during establishment of the communications link.

In addition, there are many types of error correcting codes, which are generally categorized as either convolutional codes, which correct random bit errors, and block codes, which correct burst errors. Two or more error correcting codes may be used together to obtain a total error correcting capability or power that is greater than the sum of the capabilities of the individual codes, and are typically referred to as "concatenated" codes. A popular concatenated code uses a convolutional "inner" code and a block "outer" code. The performance of a block error correcting code, moreover, may often be increased using an "interleaving" technique, in which data which may be subject to a burst error is spread out over multiple blocks, thereby providing each block code a higher probability of correcting a small portion of the large burst error. The correcting power of an interleaving technique is

determined or measured as a function of interleaver depth.
Trellis coding techniques may also be utilized.

The preferred embodiment utilizes a Reed-Solomon error correcting code for forward error correction on the upstream channels, without additional convolutional coding and interleaving. The Reed-Solomon error correcting code is known and is a block code, such that the error correcting code is computed over a block of data having a fixed size. A Reed-Solomon code is typically specified by a parameter pair (n, k), in which "n" is the code word size and "k" is the block size, such that an n-byte code word consists of k data bytes and (n - k) redundancy bytes (which represent the error correcting code information). The maximum number of symbol errors that can be corrected by a Reed-Solomon code is $t = (n - k)/2$, where a symbol is typically one 8-bit byte. A commonly used Reed-Solomon code is a (128,122) code, where the code word size is 128 bytes, each code word consists of 122 data bytes and 6 redundancy bytes, enabling a decoder to correct up to three byte errors in each 128 byte code word. In addition to a Reed-Solomon code, other error correcting codes and encryption algorithms may also be used.

In a typical prior art forward error correction implementation, the forward error correction parameters are set to a predetermined and fixed value to compensate for a particular level of noise on the communications channel. If the noise level on the communications channel becomes excessive, such that the noise exceeds the ability of the forward error correction to correct transmission errors, the data will be received in error. In that case, the data must be retransmitted or, in a worst case situation, the communications channel may no longer be usable. In either situation, data throughput is significantly decreased (or eliminated).

Forward error correction parameters, however, illustrate a balance between the amount of overhead added by the error correcting code itself (which utilizes space which could have

been used for data and therefore decreases data throughput), on the one hand, and the amount of error correction needed due to channel conditions (which may serve to increase data throughput through avoidance of retransmission), on the other hand. In the preferred embodiment, to maximize throughput of user data over a given communications channel, the optimal error correcting methodology would utilize precisely enough error correction to compensate for the existing noise level, and no more or less. Any more error correcting capability lowers throughput due to excessive overhead from transmitting the error correcting code information, while any less lowers the throughput due to the overhead caused by retransmission of data received in error. The level of noise on a communications channel, however, may vary over time, rendering a selection of a fixed set of forward error correction parameters less than optimal at any given time. One prior art method, as mentioned above, selects a fixed set of forward error correction parameters to compensate for a typical or anticipated noise level, but ceases to use the communication channel when the noise becomes excessive. This prior art method of utilizing fixed error correction code parameters is unsuitable for situations in which the number of available channels is limited, in which case it would be preferable to maintain a channel at a reduced throughput level rather than eliminate use of the channel altogether.

Secondly, another objective of the communication system of the preferred embodiment concerns minimizing the amount of throughput delay introduced by the communications equipment. Throughput delay in a polled protocol, for example, may be defined as the amount of time between the sending of a poll message prior to forward error correction encoding and the receipt of a response to the poll message following forward error correction decoding. Forward error correcting codes typically introduce such throughput delay due to the processing and computational requirements of error correction

encoding and decoding, and the amount of throughput delay is typically proportional to the error correcting power of the code. For example, the delay introduced by the interleaving/de-interleaving process is proportional to the interleaver depth, and the delay introduced by the Reed-Solomon encoding/decoding process is proportional to the code word size and number of redundancy bytes.

As discussed in greater detail below, the apparatus and method of the present invention provides a means for signaling and changing the forward error correction parameters (typically used on an upstream channel), based upon variable channel quality, such as variable noise or error levels. As a consequence, the preferred embodiment of the present invention provides a mechanism to optimize data throughput for varying noise levels (and corresponding error rates), while simultaneously providing a mechanism to decrease throughput delay as needed or as desired.

FIG. 4 is a flow chart illustrating channel monitoring and forward error correction parameter adjustment in accordance with the present invention. Beginning with start step 300, the method proceeds to receive encoded data having a first or initial degree of forward error correction, from a plurality of degrees of forward error correction, in step 305. The plurality of degrees of forward error correction result from the variable levels of correcting capability associated with various codes and with various parameters of such codes. For example, different error correcting capabilities result from the specification of different error correcting parameters such as (n, k) parameters, e.g., (128, 122), (200, 196), or (128, 124), from inclusion of different types of error correcting codes, such as concatenation of codes or inclusion of interleaving (with a specified depth), and by specification of any parameters associated with such codes. In the preferred embodiment, the initial degree of forward error correction may be established during an initial registration process, when a

secondary station establishes a communication link with a primary station. Typically in the preferred embodiment, the primary station polls individual secondary stations, where each poll message contains a secondary station identifier, an upstream channel number, and the parameters to be used for forward error correction on an upstream response. As explained with reference to FIG. 5, when the secondary station receives the poll on the downstream channel, it responds on the upstream channel designated in the poll message using the forward error correction parameters also specified in the poll message.

Continuing to refer to FIG. 4, following receipt of encoded data having an initial degree of forward error correction in step 305, channel parameters are monitored in step 310, such as monitoring packet error rates, bit error rates, noise levels (such as levels of impulse noise or ingress noise), other interference, or other parameters or factors which could be correlated with channel quality, error rates, and a desired or necessary degree of error correction capability. For example, monitoring an error rate may comprise monitoring a set of error rate parameters of a plurality of sets of error rate parameters in which the plurality of sets of error rate parameters consist of any of a plurality of combinations of a bit error rate, a packet error rate, a burst error rate, a block error rate, and a frame error rate. Next, a threshold level is determined in step 315, such as a threshold level of packet errors or bit errors. This threshold level may be predetermined, may be set at a default value, or may be adaptive and take on various values depending upon the allowable amount of transmission error. For example, under conditions in which few errors will be allowed, the threshold level may be comparatively low. In circumstances in which throughput latency may be more significant and more errors may be allowable, the threshold level may be comparatively high. Next, in step 320, the channel parameter (monitored in

step 310) is compared to the threshold level, step 320, such that if the channel parameter is not within a predetermined or adaptive tolerance or variance of the threshold level (i.e., is not equal to the threshold level plus or minus an allowable tolerance (or variance)), then the degree of forward error correction utilized will be revised, as explained below. The tolerance or variance level may be either predetermined, such as a fixed variance, or adaptive, such as a variance which is changeable over time. If the channel parameter is greater than the threshold level (plus the allowable tolerance, if any) in step 330, indicating that the channel (and its associated noise and distortions) has a comparatively high quality and is causing fewer errors than are capable of being corrected by the current error correcting code, then a revised forward error correcting parameter is selected which has a lower degree of forward error correction capability, i.e., is capable of correcting fewer errors, step 340. In such a case, a lower or lesser degree of forward error correction capability is selected to decrease the overhead associated with utilizing greater error correction capability, when such greater error correction capability is unnecessary because the channel parameter indicates fewer errors needing correction. Conversely, if the channel parameter is less than the threshold level (minus the allowable tolerance, if any) in step 350, indicating that the channel (and its associated noise and distortions) has a comparatively low or poor quality and is causing more errors than are capable of being corrected by the current error correcting code, then a revised forward error correcting parameter is selected which has a higher degree of forward error correction capability, i.e., is capable of correcting more errors, step 360. In this case, a greater or higher degree of forward error correction capability is selected to decrease the overhead associated with retransmission of an entire data packet due to excessive errors, when such greater error correction capability is

necessary because the channel parameter indicates more errors needing correction. Equivalently, depending upon the choice of channel parameter employed, such as noise levels or error rates, the comparative steps 330 and 350 may be reversed or modified. For example, if the selected channel parameter is an error rate, such that if the threshold error rate level with a selected tolerance is exceeded in step 330, then a revised forward error correcting parameter is selected which has a higher degree of forward error correction capability in step 340, with corresponding modifications of steps 350 and 360. Following steps 340 and 360, the revised forward error correction parameter is transmitted, step 370, for example, to a particular secondary station, such as secondary station 110_m. As this process may be both repeated for each connected (or active) secondary station 110_a through 110_n, and repeated over time (as conditions may vary), it is anticipated that different revised forward error correction parameters will be determined and transmitted to the different secondary stations, and over time, to the same secondary station. The various comparative steps 330 and 350 may also include variance or tolerance levels, such that forward error correction might not be revised unless the channel parameter differs from the threshold level by a predetermined amount or variance, to avoid any interruptions or delay due to comparatively minor variances and correspondingly minor changes in forward error correction capability. In addition, as the error parameters may vary over time, it may be desirable to utilize an average value of error parameters, rather than instantaneous values. The process ends (return step 380) following step 370, or when the channel parameter is equal to the threshold level, i.e., the channel parameter is not greater than the threshold level in step 330 and the channel parameter is not less than the threshold level in step 350, or is otherwise within the allowable tolerance or variance from the threshold level.

In the preferred apparatus embodiment illustrated in FIG. 2, the method illustrated in FIG. 4 may be programmed and stored, as a set of program instructions for subsequent execution, in the primary station 101 and, more particularly, in each of the processors 130 (with their associated memories 131), utilizing packet or bit error data from the forward error correction decoding performed by the corresponding receivers 135. To the extent that adjustable and dynamic forward error correction capability is necessary or desirable in the downstream direction, the method illustrated in FIG. 4 also may be programmed and stored, also as a set of program instructions for subsequent execution, in the processor 150 and memory 155 of a secondary station 110, such as secondary station 110_n illustrated in FIG. 3.

In summary, FIGs. 1 and 4 illustrate a method for adaptive forward error correction in a data communication system 100, the data communication system having a communications medium (such as 114, 115 or 116), with the communications medium having a plurality of communications channels. The method then comprises: (a) receiving encoded data over a first communications channel of the plurality of communications channels, the encoded data having a first degree of forward error correction of a plurality of degrees of forward error correction (step 305); (b) monitoring a channel parameter (such as error rate, ingress noise or impulse noise) of the first communications channel to form a monitored parameter (step 310); (c) determining a threshold level of the channel parameter (step 315); (d) comparing the monitored parameter with the threshold level (step 320); (e) when the monitored parameter is not within a variance (either predetermined or adaptive) of the threshold level, changing the first degree of forward error correction to a second degree of forward error correction of the plurality of degrees of forward error correction (steps 330, 340, 350 and 360); and (f) transmitting a forward error correction revision parameter on

a second communications channel of the plurality of communications channels, the forward error correction revision parameter corresponding to the second degree of forward error correction (step 370). The various first and
5 second communications channels, of course, may be the same or different logical or physical channels (such as time or frequency division multiplexed channels).

FIG. 5 is a flow chart illustrating forward error correction adjustment and data transmission in accordance
10 with the present invention. In the preferred embodiment, this forward error correction adjustment and data transmission methodology, for upstream data transmission, would also be implemented as a set of program instructions stored in the processor 150 and memory 155 of a secondary station 110.
15 Correspondingly, to the extent that adjustable and dynamic forward error correction capability is necessary or desirable in the downstream direction, the method illustrated in FIG. 5 also may be programmed and stored, also as a set of program instructions for subsequent execution, in a primary station
20 101, and more particularly, in its processor arrangement 120 (with associated memories), utilizing the forward error correction encoding capability of the associated transmitters 136.

Referring to FIG. 5, beginning with start step 400, data
25 having an initial degree of forward error correction (of a plurality of degrees of forward error correction) is transmitted on a communication channel, step 405. The initial degree of forward error correction may be either predetermined in a secondary station 110_n, such as currently
30 fixed in its software, or may be signaled or otherwise transmitted from a primary station 101, such as in a poll message as mentioned above. The communication channel is then monitored for reception of a revised forward error correction parameter, step 410, which may, for example, be
35 contained in a specific poll message to a secondary station or

may be broadcast in a message to all secondary stations. In addition, while the preferred embodiment utilizes signaling in a link layer through specific poll or broadcast messages, such signaling may be performed at any layer in a communication protocol, including the physical layer, the network layer, or in a software download (from a primary station 101 to a secondary station 110_n). If a revised degree of forward error correction is not received in step 415, for example, because a received poll continues to indicate the current degree of forward error correction or because a specific poll or other message containing a revised parameter was not received, then the method continues to transmit data at the then current degree (such as the initial degree) of forward error correction, step 420, and continues to monitor the communication channel for a revised parameter, returning to step 410. If a revised degree of forward error correction is received in step 415, then data will be transmitted utilizing the revised degree of forward error correction indicated by the revised forward error correction parameter, step 425, and the process ends, return step 430. As mentioned above with regard to FIG. 4, this process may be repeated for each secondary device and over time (as conditions may vary), and it is also anticipated that different revised forward error correction parameters will be received by the different secondary stations, and over time, by the same secondary station.

In summary, FIG. 5 illustrates a method for adaptive forward error correction in a data communications system 100, the data communications system 100 having a communications medium (114, 115 or 116), with the communications medium having a plurality of communications channels. The method then comprises: (a) transmitting encoded data over a first communications channel of the plurality of communications channels to form transmitted encoded data, the transmitted encoded data having an initial degree of forward error correction of a plurality of degrees of forward

error correction (step 405); (b) monitoring a second communications channel of the plurality of communications channels for a forward error correction revision parameter (step 410); (c) determining whether the forward error
5 correction revision parameter indicates a revised degree of forward error correction of the plurality of degrees of forward error correction (step 415); (d) transmitting encoded data having the initial degree of forward error correction when the forward error correction revision parameter does not indicate
10 the revised degree of forward error correction (step 420); and (e) transmitting encoded data having the revised degree of forward error correction when the forward error correction revision parameter indicates the revised degree of forward error correction (step 425). Also, as noted above, the various
15 first and second communications channels may be the same or different logical or physical channels.

Also in summary, FIGs. 2 and 4 illustrate an apparatus 101 for adaptive forward error correction in a data communications system 100, the data communications system
20 100 having a communications medium (114, 115 or 116), with the communications medium having a plurality of communications channels. The apparatus 101 then comprises, first, a channel interface 125 coupleable to the communications medium 114 for transmission of encoded data
25 on a first communications channel of the plurality of communications channels to form transmitted encoded data and for reception of encoded data on a second communications channel of the plurality of communications channels to form received encoded data, with the received encoded data having a
30 first degree of forward error correction of a plurality of degrees of forward error correction; and second, a processor arrangement 120 coupled to the channel interface 125, the processor arrangement 120 responsive through a set of program instructions to monitor a channel parameter of the
35 second communications channel to form a monitored

parameter; the processor arrangement further responsive to compare the monitored parameter with a threshold level of the channel parameter; and when the monitored parameter is not within a variance (or tolerance level) of the threshold level, the processor arrangement is further responsive to change the first degree of forward error correction to a second degree of forward error correction of the plurality of degrees of forward error correction, and to transmit via the channel interface a forward error correction revision parameter on the first communications channel, the forward error correction revision parameter corresponding to the second degree of forward error correction. As illustrated above, the apparatus may be embodied within a primary station 101, which also may be referred to as a primary device or primary transceiver. The channel interface, such as channel interface 125_n, may also further comprise a receiver and a transmitter, such as receiver 135_n and a transmitter 136_n. The processor arrangement 120 may also further comprise: a first processor coupled to the channel interface 125, such as processor 130_{n1}; a second processor coupled to the channel interface 125, such as processor 130_{n2}; and a master controller coupled to the first processor and to the second processor, such as master controller 121.

Also in summary, FIGs. 3 and 5 illustrate an apparatus for adaptive forward error correction in a data communications system 100, the data communications system 100 having a communications medium, with the communications medium having a plurality of communications channels. The apparatus comprises, first, a channel interface 160 coupleable to the communications medium 114 for transmission of encoded data on a first communications channel of the plurality of communications channels to form transmitted encoded data and for reception of encoded data on a second communications channel of the plurality of communications channels to form received encoded data; a

processor (or processor arrangement) 150 (including processor 150 with memory 155) coupled to the channel interface 160, the processor arrangement 150 responsive through a set of program instructions to set the transmitted encoded data to a
5 current (or initial) degree of forward error correction of a plurality of degrees of forward error correction, the processor arrangement further responsive to monitor the second communications channel for reception of a forward error correction revision parameter and to determine whether the
10 forward error correction revision parameter indicates a revised degree of forward error correction of the plurality of degrees of forward error correction; the processor arrangement further responsive to transmit encoded data having the current (or initial) degree of forward error
15 correction when the forward error correction revision parameter does not indicate the revised degree of forward error correction and to transmit encoded data having the revised degree of forward error correction when the forward error correction revision parameter indicates the revised
20 degree of forward error correction. As mentioned above, the processor arrangement may be processor arrangement 120, processor 150 or may also be the processor 150 coupled to the memory 155.

The forward error correction parameters carried in the
25 downstream poll in the preferred embodiment may specify the type or types of error correcting codes and the parameters for each error correcting code. The parameters for forward error correction may be specified in a variety of ways, depending upon the chosen embodiment and the types of codes to be
30 utilized. Utilizing a Reed-Solomon code, the (n, k) parameters may be directly specified, for example, using two bytes such as (128, 122). In the preferred embodiment, to decrease the overhead content in poll and other messages, a one byte parameter is utilized as an index to a table (look up table)
35 containing up to 256 variations or combinations of Reed-

Solomon (n, k) parameters. The 256 combinations of (n, k) parameters are selected on the basis of which are most likely to be utilized in the selected communications system, such as the CableComm™ System. For example, rather than
5 transmitting two bytes of information specifying (128, 122), in the preferred embodiment, a parameter consisting of one byte of information is transmitted as an index (or pointer) to a look up table stored in memory, which is then translated or converted to the selected (n, k) combination, such as (128,
10 122). Also in the preferred embodiment, forward error correction may be disabled altogether if a channel has sufficiently low noise, and each type of error correction (e.g., convolutional coding, block coding, concatenation, and interleaving) can be individually enabled or disabled utilizing a
15 specified set of operating parameters.

While carrying the forward error correction information in each downstream poll is the preferred method for the CableComm™ System, an alternative method is to use a special downstream message that is transmitted only when
20 error correcting power is revised. This eliminates the overhead of carrying this information in the downstream polls, which may be frequently sent messages. Moreover, in addition to specifying forward error correction parameters, the primary station could also specify, both initially and as may be
25 subsequently revised, analog parameters to be used for the upstream transmission, such as a modulation mode, carrier frequency, bit rate, baud rate, and bandwidth for each carrier. Improved throughput may be realized by changing the analog parameters, such as modulation mode, instead of the forward
30 error correction, where the amount of overhead that would be added by the forward error correction is greater than the throughput loss caused by a slower but more robust modulation mode. The primary station would then be able to vary the analog parameters, forward error correction, or both, in order
35 to compensate for channel impairments.

Although the preferred embodiment utilizes a poll/response protocol, the invention may also be applied to non-pollled protocols, and may use the same or different signaling techniques for dynamically adapting the types of forward error correction and the forward error correction parameters. The invention could also be used where dial-up or other lines are used as the upstream channel. Even though the dial-up line is a dedicated channel, channel characteristics over a public switched network may vary over time and also may vary from connection to connection. Thus, the dynamic adaptive forward error correction could be used to improve throughput over such a dedicated channel.

The ability to dynamically adapt the level of forward error correction for individual channels, in accordance with the present invention, provides several significant advantages. First, it allows continued use of channels that otherwise would have been vacated of all traffic in an implementation which used the prior art fixed level of forward error correction. Even though the throughput rate for that given channel is diminished due to the increased overhead of a more powerful error correcting code, the overall throughput of the communication system is increased through the utilization of an otherwise unacceptable channel. Second, the apparatus and method of the present invention allows the level of error correction to be tailored for each channel, so that a "clean" channel does not carry a greater amount of overhead or introduce a greater amount of throughput delay than that required to compensate for the specific and actual level of noise, rather than a predetermined or anticipated level of noise.

From the foregoing, it will be observed that numerous variations and modifications may be effected without departing from the spirit and scope of the novel concept of the invention. It is to be understood that no limitation with respect to the specific methods and apparatus illustrated

herein is intended or should be inferred. It is, of course, intended to cover by the appended claims all such modifications as fall within the scope of the claims. The invention is further defined by the following claims.

5

We claim:

1. A method for determining forward error correction parameters for adaptive forward error correction in a data communication system, the data communication system having a communications medium, the communications medium having
5 a plurality of communications channels, the method comprising:

(a) receiving encoded data over a first communications channel of the plurality of communications channels, the encoded data having a first degree of forward error correction
10 of a plurality of degrees of forward error correction;

(b) monitoring a channel parameter of the first communications channel to form a monitored parameter;

(c) determining a threshold level of the channel parameter;

15 (d) comparing the monitored parameter with the threshold level;

(e) when the monitored parameter is not within a variance of the threshold level, selecting a second degree of forward error correction of the plurality of degrees of forward
20 error correction; and

(f) transmitting a forward error correction revision parameter on a second communications channel of the plurality of communications channels, the forward error correction revision parameter corresponding to the second degree of
25 forward error correction.

2. The method for adaptive forward error correction in a data communications system of claim 1, wherein the plurality of degrees of forward error correction are comprised of any of
30 a plurality of combinations of parameters specifying block codes, convolutional codes, concatenated codes, and interleaving depth.

3. The method for adaptive forward error correction in a
35 data communications system of claim 1, further comprising:

(h) transmitting a revised analog parameter on a second communications channel of the plurality of communications channels, wherein the revised analog parameter is comprised of any of a plurality of combinations of parameters specifying a modulation mode, a carrier frequency, a bit rate, a baud rate, and a carrier bandwidth.

4. The method for adaptive forward error correction in a data communications system of claim 1, wherein step (e) further comprises:

(e1) when the monitored parameter as compared to the threshold level indicates that a lesser degree of forward error correction is appropriate, selecting the second degree of forward error correction having a lesser error correction capacity than the first degree of error correction; and

(e2) when the monitored parameter as compared to the threshold level indicates that a greater degree of forward error correction is appropriate, selecting the second degree of forward error correction having a greater error correction capacity than the first degree of error correction.

5. A method for revising forward error correction parameters for adaptive forward error correction in a data communications system, the data communications system having a communications medium, the communications medium having a plurality of communications channels, the method comprising:

(a) transmitting encoded data over a first communications channel of the plurality of communications channels to form transmitted encoded data, the transmitted encoded data having a current degree of forward error correction of a plurality of degrees of forward error correction;

(b) monitoring a second communications channel of the plurality of communications channels for a forward error correction revision parameter;

(c) determining whether the forward error correction revision parameter indicates a revised degree of forward error correction of the plurality of degrees of forward error correction;

(d) transmitting encoded data having the current degree of forward error correction when the forward error correction revision parameter does not indicate the revised degree of forward error correction; and

(e) transmitting encoded data having the revised degree of forward error correction when the forward error correction revision parameter indicates the revised degree of forward error correction.

6. An apparatus for determining forward error correction parameters for adaptive forward error correction in a data communication system, the data communication system having a communications medium, the communications medium having a plurality of communications channels, the apparatus comprising:

a channel interface coupleable to the communications medium for transmission of encoded data on a first communications channel of the plurality of communications channels to form transmitted encoded data and for reception of encoded data on a second communications channel of the plurality of communications channels to form received encoded data, the received encoded data having a first degree of forward error correction of a plurality of degrees of forward error correction; and

a processor arrangement coupled to the channel interface, the processor arrangement responsive through a set of program instructions to monitor a channel parameter of the second communications channel to form a monitored

parameter; the processor arrangement further responsive to compare the monitored parameter with a threshold level of the channel parameter and, when the monitored parameter is not within a variance of the threshold level, the processor
5 arrangement is further responsive to select a second degree of forward error correction of the plurality of degrees of forward error correction and to transmit via the channel interface a forward error correction revision parameter on the first communications channel, the forward error correction revision
10 parameter corresponding to the second degree of forward error correction.

7. The apparatus of claim 6, wherein the plurality of
15 degrees of forward error correction are comprised of any of a plurality of combinations of parameters specifying block codes, convolutional codes, concatenated codes, and interleaving depth.

8. The apparatus of claim 6, wherein the processor
20 arrangement is further responsive to transmit a revised analog parameter on the first communications channel of the plurality of communications channels, wherein the revised analog parameter is comprised of any of a plurality of combinations of parameters specifying a modulation mode, a
25 carrier frequency, a bit rate, a baud rate, and a carrier bandwidth.

9. The apparatus of claim 6, wherein when the monitored
30 parameter as compared to the threshold level indicates that a lesser degree of forward error correction is appropriate, the processor arrangement is further responsive to select the second degree of forward error correction having a lesser error correction capacity than the first degree of error correction; and when the monitored parameter as compared to the
35 threshold level indicates that a greater degree of forward

error correction is appropriate, the processor arrangement is further responsive to select the second degree of forward error correction having a greater error correction capacity than the first degree of error correction.

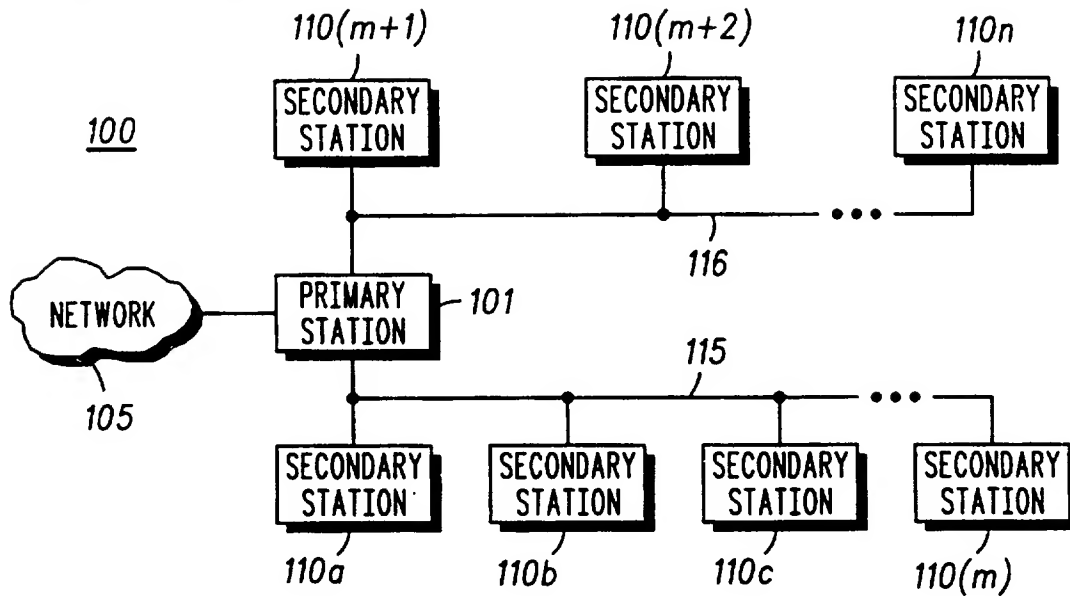
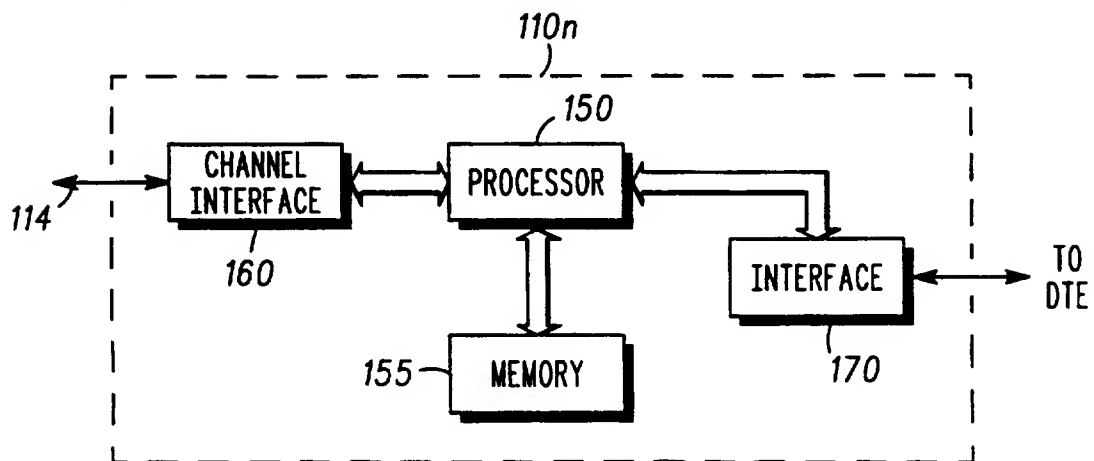
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10. An apparatus for revising forward error correction parameters for adaptive forward error correction in a data communications system, the data communications system having a communications medium, the communications medium
10 having a plurality of communications channels, the apparatus comprising:

a channel interface coupleable to the communications medium for transmission of encoded data on a first communications channel of the plurality of communications
15 channels to form transmitted encoded data and for reception of encoded data on a second communications channel of the plurality of communications channels to form received encoded data; and

a processor arrangement coupled to the channel
20 interface, the processor arrangement responsive through a set of program instructions to set the transmitted encoded data to a current degree of forward error correction of a plurality of degrees of forward error correction, the processor arrangement further responsive to monitor the second
25 communications channel for reception of a forward error correction revision parameter and to determine whether the forward error correction revision parameter indicates a revised degree of forward error correction of the plurality of degrees of forward error correction; the processor
30 arrangement further responsive to transmit encoded data having the current degree of forward error correction when the forward error correction revision parameter does not indicate the revised degree of forward error correction and to transmit encoded data having the revised degree of forward error
35 correction when the forward error correction revision

parameter indicates the revised degree of forward error correction.

FIG. 1*FIG. 3*

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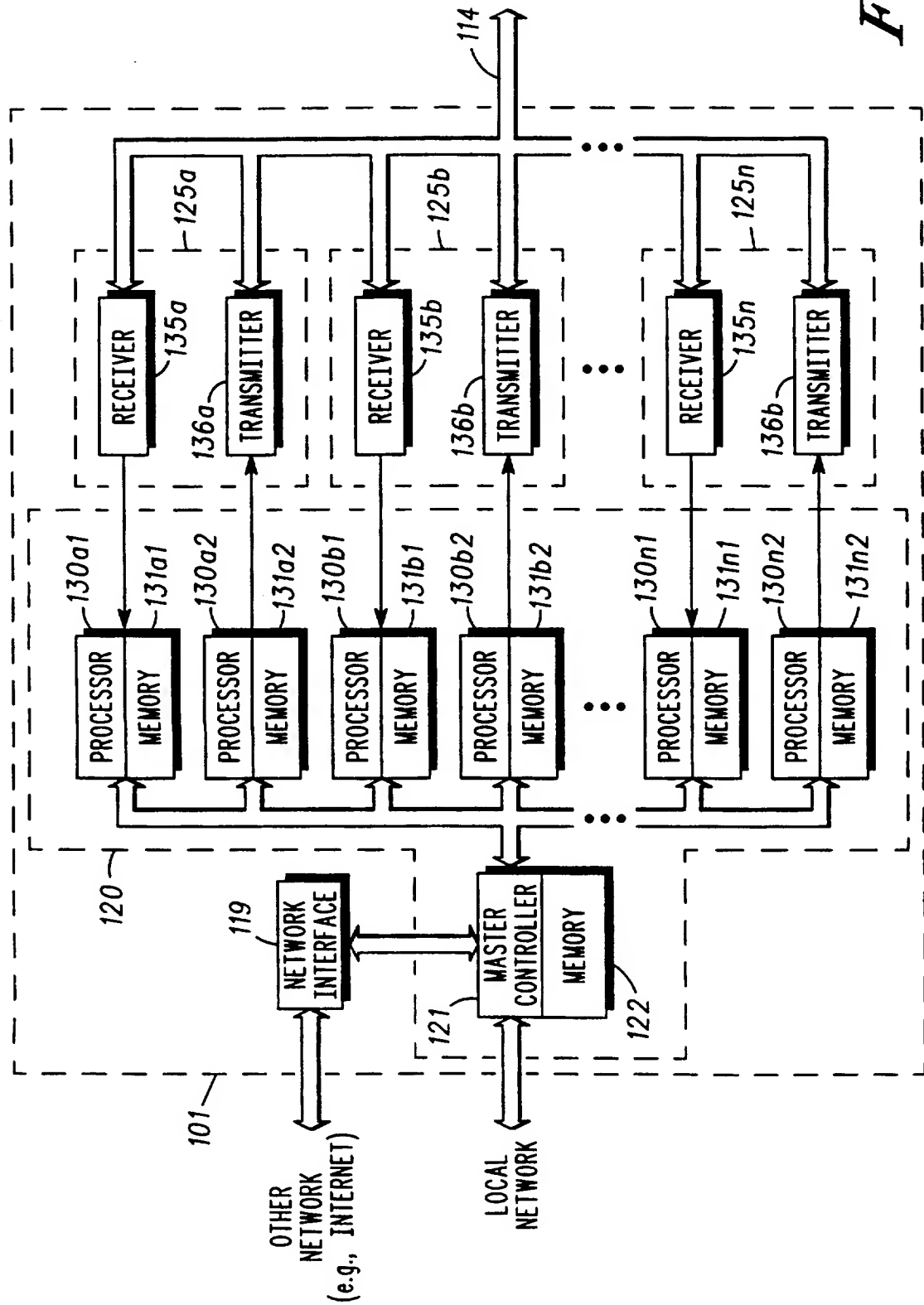


FIG. 2

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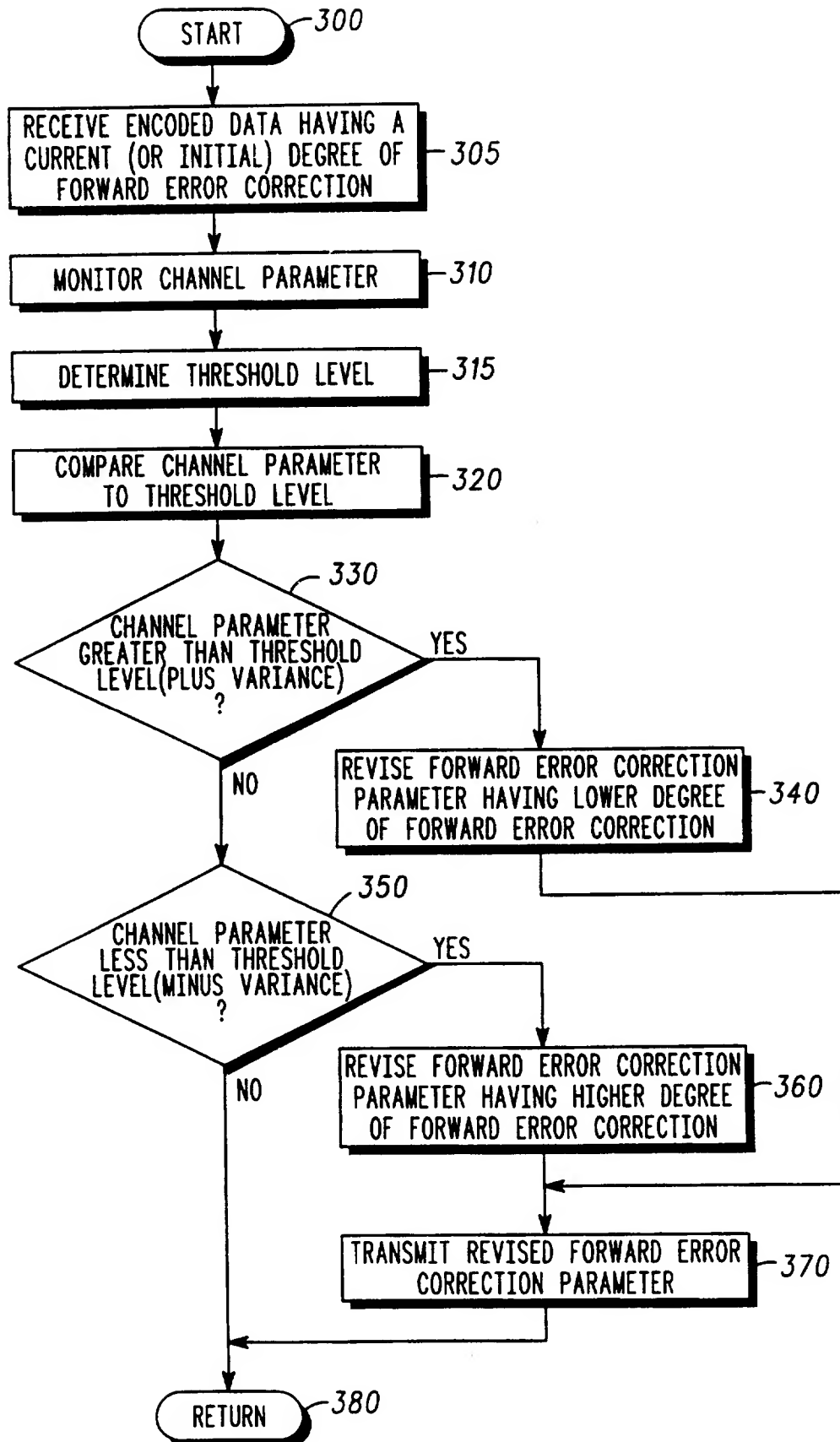
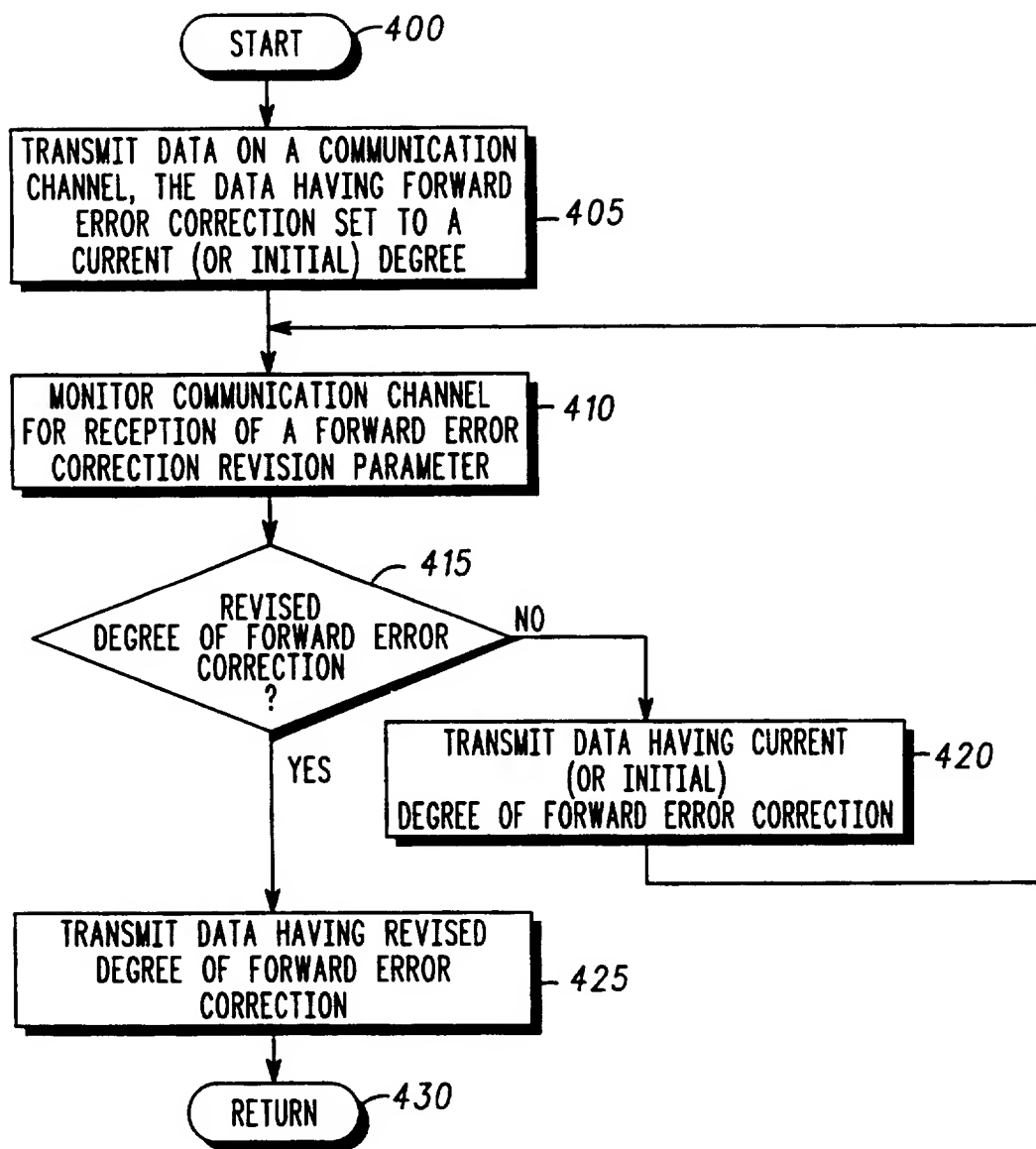


FIG. 4

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*FIG. 5*

INTERNATIONAL SEARCH REPORT

 International application No.
PCT/US97/04806

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :H03M 13/00

US CL :371/41

According to International Patent Classification (IPC) or to both national classification and IPC

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U.S. : 371/41

 Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched
NONE

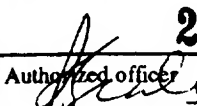
 Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
NONE

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A,P	US, 5,546,411 A (LEITCH ET AL) 13 August 1996, abstract.	1-10
A,P	US, 5,600,663 A (AYANOGLU ET AL) 04 February 1997, abstract.	1-10

☐ Further documents are listed in the continuation of Box C.
 ☐ See patent family annex.

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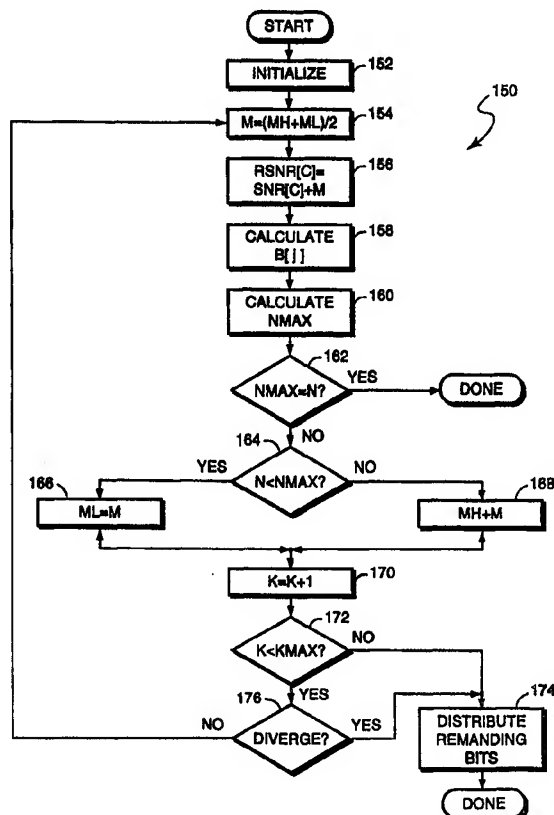


INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

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(21) International Application Number: PCT/US98/11845 (22) International Filing Date: 9 June 1998 (09.06.98) (30) Priority Data: 08/873,421 12 June 1997 (12.06.97) US (71) Applicant: AWARE, INC. [US/US]; 40 Middlesex Turnpike, Bedford, MA 01730 (US). (72) Inventors: TZANNES, Marcos; Unit #53, 665 Lowell Street, Lexington, MA 02173 (US). KECHRIOTIS, George; 34 Hamilton Road #2, Arlington, MA 02174 (US). WU, Pui; 41A Tremont Street, Malden, MA 01248 (US). (74) Agents: O'DONNELL, Martin, J. et al.; Cesari and McKenna, LLP, 30 Rowes Wharf, Boston, MA 02110 (US).		(81) Designated States: AL, AU, BA, BB, BG, BR, CA, CN, CU, CZ, EE, GE, GW, HU, ID, IL, IS, JP, KP, KR, LC, LK, LR, LT, LV, MG, MK, MN, MX, NO, NZ, PL, RO, SG, SI, SK, SL, TR, TT, UA, UZ, VN, YU, ARIPO patent (GH, GM, KE, LS, MW, SD, SZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, ML, MR, NE, SN, TD, TG). Published <i>With international search report.</i> <i>Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.</i>

(54) Title: ADAPTIVE BIT ALLOCATION FOR VARIABLE BANDWIDTH MULTICARRIER COMMUNICATION**(57) Abstract**

Data is distributed among the channels of an asynchronous data subscriber loop (ADSL) communications system in accordance with an adaptive algorithm which from time to time measures the signal to noise ratio of the various channels and finds a margin for each channel dependent on achievement (where possible) of a given bit error rate and a desired data transmission rate. The margin distribution is achieved by augmenting the constellation signal to noise ratio to enhance computational efficiency and allow redetermination of bit allocation tables during transmission as necessary. Pairs of bit allocation tables are maintained at the transmitter and receiver and one table of each pair at the transmitter and receiver is updated while the other pair is in use for controlling communication.



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ADAPTIVE BIT ALLOCATION FOR VARIABLE BANDWIDTH MULTICARRIER COMMUNICATION

TECHNICAL FIELD

This application relates to the field of electronic communication and more particularly to the field of multiband digital signal communication.

BACKGROUND OF THE INVENTION

Conventional multicarrier digital communication is a technique for transmitting and receiving digital signals using a plurality carriers (subchannels) having different frequencies. Each of the subchannels is used to communicate a different portion of the signal. The transmitter divides the signal into a number of components, assigns each component to a specific one of the carriers, encodes each of the carriers according to the component assigned thereto, and transmits each of the carriers. The receiver decodes each received carriers and reconstructs the signal.

The maximum amount of information that can be encoded onto a particular subcarrier is a function of the signal to noise ratio of the communication channel with respect to that subcarrier. The signal to noise ratio of a communication channel can vary according to frequency so that the maximum amount of information that can be encoded onto one carrier may be different than the maximum amount of information that can be encoded onto another carrier.

Bit loading is a technique for assigning bits to subchannels according to each subchannel's signal to noise ratio. A bit loading algorithm provides a bit allocation table that indicates the amount of information (in bits) that is to be encoded on each of the carriers. That is, for a multicarrier communication system with J carriers, a bit allocation table $B[j]$ indicates, for each $j = 1$ to J , the amount of information that is to be encoded onto each of the J carriers.

Shaping the transmission to match the channel characteristics is known. For example, a technique known as "water pouring" was introduced by Gallager in 1968

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(“Information Theory and Reliable Communication”, page 389) and by Wozencraft in 1965 (“Principles of Communication Engineering”, pp. 285-357). Water pouring involves distributing the energy of the transmission signal according to the channel frequency response curve (a plot of the signal to noise ratio as a function of frequency).
5 The frequency response curve is inverted and the available signal energy (the “water”) is “poured” into the inverted curve so that more of the energy is distributed into those portions of the channel having the highest signal to noise ratio. In a multicarrier system in which the transmission band is divided into numerous subchannels, throughput can be maximized by putting as many bits in each subcarrier as can be supported given the
10 “water pouring” energy and a desired error rate.

Other techniques for allocating bits among carriers of a multicarrier signal are known. U.S. Patent No. 4,731,816 to Hughes-Hartogs discloses a bit loading scheme where one bit at a time is incrementally added to each subcarrier until a maximum rate is achieved. Subcarriers that require the least amount of additional power to support an
15 additional bit are selected first.

U.S. Patent No. 5,479,477 to Chow et al. discloses a bit loading scheme that is capable of either maximizing the throughput or maximizing the margin for a particular target data rate. Unlike Hughes-Hartogs, Chow et al. determines the bit loading table one carrier at a time (rather than one bit at a time). In Chow et al., all the carriers are
20 sorted in descending order according to the measured signal to noise ratio. The initial subchannels that are selected are the ones capable of carrying the most bits. Using the Chow et al. scheme to maximize the data rate provides a bit loading table similar to that provided by the Hughes-Hartogs algorithm.

In order for the receiver to correctly interpret the received data, both the
25 transmitter and the receiver must use the same bit loading table. When the bit loading algorithm is performed during the initialization phase of communication, the resulting bit allocation table is communicated between the transmitter and receiver to ensure that both the transmitter and the receiver are using the same bit loading table. However, in the event that the communication channel signal to noise ratio characteristics change
30 during communication, it may be necessary to update/change the bit allocation table to more appropriately match the transmission with the channel characteristics. However,

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when the bit allocation table is changed, it is necessary to synchronize use of the new table with both the transmitter and the receiver. If the transmitter and the receiver use different bit allocation tables at any time, the communications link will suffer significant errors in those subchannels in which the bit allocation tables do not agree.

5 In addition, determining a new bit allocation table can be time consuming, especially if the bit loading algorithm is computationally intensive, such as that disclosed by Hughes-Hartogs where the bit allocation table is constructed one bit at a time. If the bit allocation table is to be calculated many times during communication between the transmitter and receiver, then spending a relatively long amount of time recalculating the
10 bit allocation table (and hence not communicating data) is undesirable.

One solution is to simply not change the bit loading table after initialization. However, this may be unacceptable in cases where the communication channel signal to noise ratio changes during data transmission. Accordingly, it is desirable to be able to determine a bit loading table relatively quickly and to be able to synchronize use of the
15 new table by the transmitter and the receiver.

SUMMARY OF THE INVENTION

In accordance with the present invention, a pair of bit allocation tables are maintained at both the transmitter and the receiver. These tables are updated as needed, using measurements of the signal to noise ratio performed on known data transmitted to
20 the receiver in a control frame separate from the data frame. The transmitter signals the receiver as to which of the two tables is to be used for subsequent communication. Preferably, this is done by transmitting a flag from the transmitter to the receiver at some point during the data transmission; this causes the receiver to thereafter switch the bit loading table it is using for communication to synchronize with the corresponding table
25 at the transmitter.

In the preferred embodiment of the invention, although the invention is not restricted thereto, 69 "frames" of 245.5 microseconds duration each are used to form a "superframe" of 16.94 milliseconds. The first frame of each superframe comprises a control frame that is used to transmit a standard (known) data set from the transmitter
30 to the receiver; the remaining frames contain data. The receiver measures the signal to

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noise ratios of the received data in this frame for each of the channels and uses this to calculate channel bit allocations for subsequent data transmissions. In practice, it has not been found necessary to calculate the signal to noise ratios for each and every super-frame, although this can, of course, be done. Rather, we have found it sufficient for most data transmissions to measure the signal to noise ratios of the channels over several frames, average them, update the bit allocation tables based on the resultant values, and use the bit allocations tables so determined over hundreds or thousands of subsequent frames.

The bit allocation table updating is performed by comparing the measured signal to noise ratio (SNR) in each channel with a constellation signal to noise ratio $\text{SNR}[c_j]$, that has been augmented by a trial noise margin M , $\text{SNR}_a[c_j] = \text{SNR}[c_j] + M$. The constellation signal to noise ratio, $\text{SNR}[c_j]$, specifies the number of bits c_j ("constellation size") that can be transmitted over a channel j given a specific signal to noise ratio SNR_j , where c_j may vary, for example, from 1 to 15. The value of the margin M is dependent on the difference between the amount of data (i.e., number of bits) that can be transmitted across the channels in accordance with the augmented constellation signal to noise ratio $\text{SNR}_a[c_j]$ and the amount that is desired to be transmitted (the "target data rate"), N . The value of this margin is varied in order to optimize it for the particular communication conditions as manifested by the measured signal to noise ratios, SNR_j .

In particular, the total number of bits that may be transmitted over J channels, each characterized by signal to noise ratio SNR_j , is $N_{\max} = \sum_{j=1}^J c_j$, where the respective

c_j are determined from the measured signal to noise ratios, SNR_j . See, for example, "Digital Communications" by John G. Proakis, pp. 278ff for channel capacity calculations for quadrature amplitude modulation (QAM) systems, the preferred form of transmission for this invention. Preferably, the channel capacity calculations are performed in advance and stored in the form of lookup tables for rapid access. In the preferred embodiment described herein, the margin M is determined as $M = (10/J)^* (N_{\max} - N)$. The augmented constellation signal to noise ratio is then given by

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$SNR_a[c_j] = SNR[c_j] + M$, and this value is used to determine (e.g., by table lookup as described above) the number of bits that can be transmitted over a channel. By augmenting the constellation signal to noise ratio, $SNR[c_j]$, rather than the channel signal to noise ratio, SNR_j , fewer additions are required, since the range of constellation sizes (e.g., $c_j = 1 \dots 15$) is typically smaller than the range of channels (e.g., $j = 1 \dots 256$).

As long as the amount of data that can be transmitted over the channels in a given interval differs (as determined by the calculations just described) from the amount of data desired to be transmitted in that interval, i.e., $N_{max} \neq N$, and assuming that certain other exit conditions have not been satisfied, the receiver cycles through a loop that repeatedly adjusts the margin M and recalculates N_{max} . To do this, the receiver sets a high margin threshold M_H and a low margin threshold M_L . During those superframes in which the bit allocation table is to be recalculated, the high threshold and low threshold margins are initialized to either a first state ($M_H = 0$, $M_L = (10/J) * [N_{max} - N]$) or a second state ($M_L = 0$, $M_H = (10/J) * [N_{max} - N]$) dependent on whether N_{max} is greater than N or less than N .

Thereafter, in each iteration, either the high or the low margin is adjusted in the search for the condition in which $N_{max} = N$. Specifically, at the beginning of subsequent (non-initialization) iterations, the margin is set to the average of the high and low margin thresholds, $M = (M_H + M_L)/2$, and the augmented constellation signal to noise ratio $SNR_a[c_j]$, the bit allocation table $B[j]$, and the calculated capacity N_{max} are determined.

If the calculated capacity exceeds the desired capacity, i.e., $N_{max} > N$, the receiver increases the low margin threshold margin to M , i.e., it sets $M_L = M$. If the calculated capacity is less than the desired capacity, i.e., $N_{max} < N$, the receiver decreases the high threshold, i.e., it sets $M_H = M$. The iteration then repeats.

The receiver exits from the loop on the occurrence of any of several conditions. A first occurs when it is determined that $N_{max} = N$. This is the desired solution, and represents an optimum equal distribution of margin over the communication channels.

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A second occurs when the test condition ($N_{\max} - N$) is diverging. A third occurs when the desired equality is not achieved after a defined number of iterations. In one system implemented according to the preferred embodiment described herein, we have found a limit of 16 iterations sufficient.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a schematic diagram of an ADSL communications system showing bit allocation tables in accordance with the present invention;

Fig. 2 is a diagram of control and data frames as used in connection with the present invention;

Fig. 3 is a graph illustrating a multicarrier communication system.

Fig. 4 is a graph illustrating signal-to-noise ratio as a function of frequency.

Fig. 5 is a graph illustrating bit loading and margin for a multicarrier communication system.

Fig. 6 is a flow chart illustrating a bit loading algorithm for a multicarrier communication system.

Fig. 7 is a flow chart illustrating initialization for the bit loading algorithm of Fig. 6.

Fig. 8 is a flow chart illustrating operation of a receiver software for calculating, modifying, and synchronizing a change in a bit allocation table used in a multicarrier communication system.

DETAILED DESCRIPTION OF AN ILLUSTRATIVE EMBODIMENT

In Figure 1, a transmitter 10 for use in asynchronous data subscriber loop (ADSL) communications has first and second bit allocation tables 12 and 14 for use in assigning

data to a plurality of channels for transmission to a remote receiver 16 which has corresponding bit allocation tables 20 and 22. The tables operate in pairs under control of a table controller 24 at the transmitter. In accordance with ADSL practice, a digital signal $s(t)$ to be transmitted to a receiver is distributed over a plurality of channels f_1, f_2, \dots, f_j , in accordance with channel allocation assignments stored in the bit allocation tables.

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In particular, the tables $B[j]$ define, for each channel j , the number of bits that can reliably be transmitted over a particular channel at a given bit error rate at the specific signal to noise ratio measured for that channel. These tables are determined as described in detail herein, and may vary from time to time during the course of a transmission.

5 At any given time, a single table, e.g., table 12, is used for transmission at the transmitter, and a corresponding table, e.g., table 20, is used for reception at the receiver. These tables are images of each other, i.e., contain the same data, and are used in pairs, so that reliable communication can occur. Similarly, tables 14 and 22 are images of each other and are used in pairs.

10 A table control unit 24 at the receiver controls the formation of the bit allocation tables 12, 14, 20, and 22. It measures the signal to noise ratio on each of the channels f_1, f_2, \dots, f_j , compares the measured values with predetermined values defining the bit capacity of a channel at given signal to noise values, augmented with noise margins as described herein, and thus determines the bit allocation for each channel. The allocations so defined are stored in the tables 20 and 22 at the receiver. They are also transmitted back to the transmitter, e.g., via a control channel 26, and are there stored as the tables 12 and 14, respectively. After initial loading, the transmission is advantageously arranged such that only updated tables are transmitted back to the transmitter.

20 At the transmitter 10, a table switch unit 28 selects which of the two table pairs (12, 20; 14, 22) are to be used in a given transmission and reception. Typically, a given pair will continue in use until the communication conditions change sufficiently that the bit allocations among the channels change. At that time, a new table must be formed at the receiver, and communicated to the transmitter. When this occurs, the table switch unit 28 typically will switch to the new table for subsequent transmissions. When it does so, it transmits a flag to the receiver that indicates that a switch to the alternative pair is to take place. This switch will usually be made effective as of the next superframe, but may, by prearrangement with the receiver, be made effective at some agreed upon point after that.

30 Fig. 2 is a diagram of a superframe 30. It is formed from a control frame 32 and a number of data frames 34. During the control frame interval, the transmitter sends to the receiver a known signal from which the receiver can measure the signal to noise ratio

of each of the channels in order to calculate the bit allocations. The remaining frames of the superframe comprise data frames for the transmission of the desired data. In a preferred embodiment of the invention, there are one control frame and 68 data frames, each of 245.5 microsecond duration, for a superframe time of 16.94 milliseconds.

5 Referring to Fig. 3, a graph 100 illustrates multicarrier signal transmission. The graph 100 has a horizontal axis 102 representing frequency wherein lower frequencies are toward the left side of the axis 102 while higher frequencies are toward the right side of the axis 102. The graph 100 illustrates that a multicarrier signal, incorporating J discrete carrier signals, is transmitted via carriers at frequencies f_1, f_2, \dots, f_j .

10 Each of the carriers shown in the graph 100 is capable of transmitting a certain number of bits of information. Accordingly, the total number of bits transmitted via the multicarrier signal is the sum of the number of bits that can be transmitted by each of the carriers. For example, if each of the carriers can transmit three bits of information, then the signal shown in the graph 100 can transmit a total of $J*3$ bits of information.

15 In a preferred embodiment, each of the carriers transmits information using quadrature amplitude modulation (QAM), a conventional digital signal encoding technique where different combinations of amplitude and phase of each carrier signal represent different digital values. For example, a carrier signal can be encoded using two different possible amplitudes (A1 and A2) and two different possible phases (P1 and P2)
20 so that the carrier can represent one of four possible values: a first value when the carrier signal has amplitude A1 and P1, a second value corresponding to a combination A1 and P2, a third value corresponding to a combination A2 and P1, and a fourth value corresponding to a combination A2 and P2. The various combinations of amplitude and phase for a given carrier signal is called a "constellation". Note that the number of bits
25 that can be transmitted via a particular carrier is a function of the maximum possible constellation size for that carrier.

For each carrier, the maximum size of the constellation, and hence the maximum number of bits that can be transmitted via that carrier, is a function of the signal to noise ratio (SNR) of the communication channel and is a function of the desired bit error ratio (BER). The BER is the number of single bit transmission/reception errors per the total
30

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number of bits transmitted. Increasing the number of discrete amplitudes and/or phases associated with a particular carrier (i.e., increasing the constellation size) increases the likelihood of bit errors. The BER increases with increasing constellation size because, as the number of discrete amplitudes and/or phases increases, the magnitude of the difference between discrete phases and/or amplitudes decreases and hence the ability of the receiver to distinguish between different phase and/or amplitude values decreases.

The relationship between BER and SNR is well-known in the art of multicarrier communication. Tables are available that show the minimum SNR that can support a BER of a fixed amount or less for a given constellation size. For example, the table shown below, SNR[c_j], a constellation signal to noise ratio, indicates the minimum SNR needed to transmit a constellation having the indicated size in order to obtain an expected BER of 10^{-7} (i.e., an error of one bit per every 10^7 bits that are transmitted.) Note that as the constellation size increases, the minimum required SNR also increases.

	Constellation size c (in bits)	SNR requirement
15	2	14 dB
	3	19 dB
	4	21 dB
	5	24 dB

Referring to Fig. 4, a graph 110 illustrates a relationship between SNR and frequency for a communication channel transmitting a multicarrier signal having carriers between frequencies f_1 and f_j . A vertical axis 112 of the graph 110 represents SNR. A horizontal axis 114 of the graph 110 represents frequency in a manner similar to that illustrated in connection with the horizontal axis 102 of the graph 100 of Fig. 3.

A plot 116 shows the relationship between SNR and frequency for the frequencies between f_1 and f_j , the lowest and highest (respectively) carrier frequencies for the multicarrier frequency signal. The plot 116 illustrates that the SNR varies according to frequency so that, for example, the SNR at frequency f_m is lower than the SNR at frequency f_n . Based on the table shown above, it is possible that, for a given BER, the

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constellation size supported by the carrier frequency f_m is smaller than the constellation size supported by the carrier frequency f_n .

Referring to Fig. 5, a graph 120 uses a plot 122 to illustrate a hypothetical relationship between SNR and frequency. The graph 120 is similar to the graph 110 of Fig. 4. The vertical axis of the graph 120, which represents SNR, has superimposed thereon the SNR requirement numbers from the table, shown and discussed above, that relates minimum SNR requirements with constellation size for a BER of 10^{-7} . The graph 120 shows that an SNR of 14 dB is required to support a constellation size of two bits and that SNR's of 19, 21, and 24 are required to support constellation sizes of three, four, and five bits, respectively. Based on this, it is possible to use the plot 122 to determine a maximum constellation size for each of the carrier frequencies between f_1 and f_j . For example, the plot 122 shows that any carrier frequencies between f_1 and f_a can support a maximum constellation size of four bits since all portions of the plot 122 between f_1 and f_a are greater than 21 dB (the minimum required SNR to support a constellation size of four bits), but less than 24 dB (the minimum SNR for five bits). No carrier frequencies between f_1 and f_a can support a constellation size of five bits at the BER used to generate the minimum SNR requirements.

The portion of the plot 122 between f_a and f_b is shown in Fig. 5 as being greater than 24 dB. Accordingly, carrier frequencies between f_a and f_b can support a maximum constellation size of at least five bits. Similarly, carrier frequencies between f_b and f_c will support a maximum constellation size of four bits; carrier frequencies between f_c and f_d will support a maximum constellation size of three bits; carrier frequencies between f_d and f_e will support a maximum constellation size of two bits, and carrier frequencies between f_e and f_j will support a maximum constellation size of three bits.

The difference between the minimum required SNR and the actual transmission channel SNR is called the "margin". For example, the plot 122 shows that if four bits are used at the carrier frequency f_1 , the carrier frequency at f_1 will have a margin somewhat greater than zero since the SNR at f_1 is shown in Fig. 5 as being greater than

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the minimum SNR requirement of 21 dB. Similarly, it is possible to use less than the maximum supported constellation size at a particular carrier frequency. For example, although the plot 122 shows that a carrier at the frequency f_a will support a constellation size of five bits (since the SNR at f_a is 24 dB), it is possible to encode the carrier at the frequency f_a with only three bits. In that case, the margin at the frequency f_a is the difference between the transmission channel SNR at f_a (24 dB) and the SNR required to support a constellation of three bits at frequency f_a (19 dB). Accordingly, the margin at frequency f_a is 5 dB.

In instances where the multicarrier signal is used to transmit the maximum number of data bits, then the SNR of the communication channel is first measured and then each carrier is set to the maximum supported constellation size. However, in many applications, the multicarrier signal is used to transmit less than the maximum possible number of bits. In those cases, it is advantageous to maximize the overall margin of the signal to thus reduce the error rate. This can be illustrated by a simple example:

Assume a two-channel multicarrier signal has a maximum constellation size of five bits for the first carrier and four bits for the second carrier. Further assume that it is desirable to use the signal to transmit six bits. One way to allocate the bits among the two carriers is to use the first carrier to transmit five bits and the second carrier to transmit one bit. In that case, however, the margin for the first carrier is relatively small while the margin for the second carrier is relatively large. There will be many more errors for bits transmitted via the first carrier than bits transmitted via the second carrier and, since most of the bits are being transmitted via the first carrier anyway, then the overall error rate of the signal, while below the target BER, is still higher than it has to be in this case. A more advantageous way to allocate the bits might be to allocate three bits to each of the two carriers. In that case, both of the carriers operate with a relatively large margin and the overall error rate of the signal is reduced.

Of course, in many multicarrier communication applications, there are hundreds of carriers and hundreds to thousands of bits that are transmitted. In addition, it is necessary to allocate the bits in a relatively rapid manner since time spent allocating bits is time not spent communicating information. Furthermore, it may be necessary to reallo-

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cate the bits during communication if the channel transmission characteristics change dynamically.

Referring to Fig. 6, a flow chart 150 illustrates a technique for allocating bits among carriers of a multicarrier signal. Processing begins at a first step 152 where various quantities used to allocate the bits are initialized. These quantities include M_H , the high-bound for the margin, M_L , the low-bound for the margin, and k , an iteration counter which is described in more detail below. Following step 152 is a step 154 where the margin, M , is calculated by averaging M_H and M_L .

Following step 154 is a step 156 where a table indicating required SNR for various constellation sizes, $RSNR[c]$, is calculated. $RSNR[c]$ is a table having entries equal to the sum of the margin, M , and the minimum SNR requirements that can support a constellation of size c , and thus comprises an augmented constellation signal to noise ratio, $SNRa[c_j] = SNR[c_j] + M$. Following step 156 is a step 158 where a bit table, $B[j]$, is calculated. $B[j]$ is a table of the maximum number of bits that can be allocated to each of the carriers f_1, \dots, f_j , given the values stored in $RSNR[c]$. The maximum number of bits are allocated for each carrier in a manner similar to that discussed above in connection with Fig. 5.

Following step 158 is a step 160 where a value N_{max} is calculated. N_{max} represents the maximum number of bits that can be transmitted on the channel and is determined by summing all of the values in the table $B[j]$. Since the table $B[j]$ contains the maximum number of bits that can be transmitted for each carrier based on the minimum required SNR for each constellation size plus the calculated margin, then N_{max} represents the maximum number of bits that can be transmitted on the channel wherein each of the carriers has a margin of at least M .

Following the step 160 is a test step 162 which determines if N_{max} equals N where N is the number of bits that are to be transmitted using the multicarrier signal. If N_{max} does in fact equal N , then processing is complete and the bit table $B[j]$ represents an allocation of bits among the carriers of the multicarrier signal wherein each carrier will have a margin at least as large as M .

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If it is determined at the test step 162 that N_{\max} does not equal N , then processing transfers from the test step 162 to a test step 164. Note that if N is less than N_{\max} , then the margin can be increased (in order to decrease N_{\max}) in the next iteration. Similarly, if N is not less than N_{\max} , then the margin is too large and needs to be decreased in the next iteration. If it is determined at the test step 164 that N is less than N_{\max} , the control transfers from the test step 164 to a step 166 where M_L , the low-bound on the margin, is set equal to M . Setting M_L equal to M effectively increases M_L , causing an increase in the value of the margin, M , that will be calculated on the next iteration at the step 154.

Conversely, if it is determined at the step 164 that N is not less than N_{\max} , then control transfers from the step 164 to a step 168 where M_H , the high-bound on the margin is set equal to M . This effectively decreases the value of M_H , thus causing the value of M to decrease when M is calculated at the step 154 on the next iteration.

Control transfers from either the step 166 or step 168 to a step 170 where the iteration counter, k , is incremented. Following the step 170 is a test step 172 which determines if the iteration counter is less than the maximum allowable value for the iteration counter, K_{\max} . The iteration counter, k , is used to ensure that the algorithm will terminate after a certain number of iterations even if the terminating condition at the step 162 (i.e., $N_{\max} = N$) is never met. In a preferred embodiment, K_{\max} equals 16.

If it is determined at the test step 172 that k is not less than K_{\max} , then control transfers from the step 172 to a step 174 where the remaining bits are either removed or added to the bit table, $B[j]$, as appropriate. Bits are added or removed at the step 174 in a random or pseudo random manner so that the sum of all allocated bits in the table $B[j]$, equals N , the number of bits that are to be transmitted via the multichannel signal. Note that in this instance, there is no guarantee that each of the carriers has a margin of at least M . The step 174 is simply executed in order to finalize the allocation process if the algorithm is unable to meet the termination condition at the step 162.

If it is determined at the test step 172 that the iteration counter, k , is less than the predetermined maximum value for the iteration counter, then control transfers from the

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step 172 to a test step 176 which determines if the algorithm is diverging, i.e., if ($N_{\max} - N$) is increasing. It is desirable for the algorithm to converge so that the value of N_{\max} gets closer to the value of N with each iteration because the algorithm terminates when N_{\max} equals N at the test step 162. However, if it is determined at the test step 176 that
5 the value of N_{\max} is actually getting farther from the value of N with each iteration, then control transfers from the step 176 to the step 174 where the remaining bits are distributed randomly among the values in the table $B[j]$, as discussed above, after which processing is complete.

If it is determined at the test step 176 that the algorithm is not diverging, then
10 control transfers from the step 176 back to the step 154 where the margin is calculated for the next iteration. The margin calculated at the subsequent iteration 154 will either be less than or greater than the margin calculated on the previous iteration, depending upon whether N was less than N_{\max} or not at the test step 164, as discussed above.

Referring to FIG. 7, a flow chart 180 illustrates in detail the initialization routine
15 for the step 152 of the flow chart 150 shown in FIG. 6. The initialization routine is entered and processing begins at a step 182 where the transmission characteristics of the channel are measured to determine the signal-to-noise ratio at each of the carrier frequencies of the multicarrier signal. As discussed above in connection with Fig.'s 4 and 5, the transmission channel signal-to-noise ratio may be a function of frequency. Meas-
20 uring the channel transmission characteristics at the step 182 is discussed in more detail hereinafter.

Following the step 182 is a step 184 where the minimum required signal-to-noise ratio table, $SNR[c]$, is initialized. As discussed above, for a given bit error ratio (BER), the minimum required SNR for each constellation size, c , can be determined via conven-
25 tional calculations known in the art or by looking up the values in a textbook. Following the step 184 is a step 186 where the bit table, $B[j]$, is calculated. Calculation of the bit table at the step 186 is similar to calculation of the bit table at the step 158 discussed above in connection with the flow chart 150 of FIG. 6, except that the unaugmented SNR table is $SNR[c_j]$ used at the step 186 rather than the RSNR table which is used at
30 the step 158. Using the SNR table at the step 186 effectively calculates the bit table,

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$B[j]$, with a margin of zero. Following the step 186 is a step 188 where N_{\max} is calculated. The step 188 is similar to the step 160 discussed above in connection with the flow chart 150 of FIG. 6; N_{\max} is simply the sum of all the entries in the bit table, $B[j]$.

Following the step 188 is a step 190 where it is determined if N_{\max} equals N . If N_{\max} does equal N at the step 190, then processing is complete for the entire algorithm (not just the initialization portion) since the channel will only support N_{\max} bits of transmission. That is, if N_{\max} equals N at the step 190, there is no point in continuing with the algorithm and calculating a margin since, by default, the channel can transmit no more than N bits.

If it is determined at the test step 190 that N_{\max} does not equal N , then control transfers from the step 190 to a test step 192 where it is determined if N is less than N_{\max} . Note that if N is not less than N_{\max} at the step 192, then the channel will not support transmission of N bits at the BER used to construct the SNR table at the step 184. That is, the bandwidth of the channel is too low. However, in this case, the algorithm can continue by calculating a negative margin and simply proceeding to maximize the negative margin so that, although the BER that will be achieved will exceed the desired BER, it is still minimized given the requested data rate. In another embodiment, the algorithm can terminate at this point and indicate that the bits cannot be allocated. In yet another embodiment, the algorithm can be rerun using a higher BER and (presumable) lower minimum SNR requirements for the various constellation sizes.

If it is determined at the step 192 that N is not less than N_{\max} (i.e., the system will be operating with a negative margin) then control transfers from the step 192 to a step 198 where the low-bound on the margin M_L , is set to zero. Following the step 198 is a step 200 where the high-bound on the margin is set using the formula $M_K = (10/J^*)(N_{\max} - N)$. Note that, however, in this case the high-bound on the margin will be set to a positive value at the step 200 because $N_{\max} - N$ will be a positive number.

Following either the step 200 or the step 196, control transfers to a step 202 where the iteration counter that is used to terminate the algorithm after a predetermined number of iterations is set to one. Following the step 202, the initialization routine is

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exited so that the remaining processing, as discussed above in connection with FIG. 6, can continue.

The formula used to set M_L at the step 196 and to set M_H at the step 200 provides upper and lower bounds of the margin such that the algorithm converges in a reasonable number of iterations while ensuring that the final margin does not fall outside the range between M_L and M_H . Of course, it is possible to practice the invention using other formulas or techniques for calculating initial values for M_L and M_H .

Referring to FIG. 8, a flow chart 210 illustrates operation of software used by the receiver to allocate bits among the different carriers of the multicarrier signal and synchronize changes in the bit allocation table with the transmitter. Processing begins at a first test step 262 which determines if the receiver has received a reference frame. A reference frame is a predetermined and detectable frame of special data bits that is provided by the transmitter to the receiver to allow the receiver to determine the channel characteristics. In a preferred embodiment, the reference frame is transmitted periodically, although other conventional techniques can be used to determine whether the reference frame should be sent by the transmitter. The reference frame is recognized by the receiver using any one of a variety of conventional techniques such as a special header in a packet indicating that a reference frame is being provided. Use of a reference frame in connection with multicarrier communication is well-known in the art. If a reference frame is not received at the step 262, the software loops back to the test step 262 to poll for receipt of the reference frame.

If it is determined at the test step 262 that a reference frame has been received, then control transfers from the step 262 to a step 264 where the errors in the reference frame are measured with respect to the brown constellation distances of the first signal. Note that since the reference frame is a predetermined signal, the receiver can know exactly what was sent by the transmitter. Therefore, any differences between the data received by the receiver and the expected values for signal data can be accounted for by errors induced by the transmission channel. These errors are measured at the step 264.

Following the step 264 is a step 266 where the receiver determines the channel characteristics based on the errors measured at the step 264. This is done in a conventional manner using techniques for determining channel characteristics based on detected

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transmission errors. Following the step 266 is a step 268 where the receiver allocates various bits among the carriers using, in a preferred embodiment, the technique disclosed above in connection with Fig.'s 6 and 7.

Following the step 268 is a test step 270 which determines if the bit allocation
5 table provided at the step 268 is different than the previous bit allocation table. That is, it is determined at the step 270 if there is a difference between the recently-calculated bit allocation table and the previous bit allocation table. If it is determined at the test step 270 that there is no difference (i.e., that the bit allocation table has not changed), then control transfers from the step 270 back to the step 262 where the software waits for the
10 transmitter to send another reference frame. Otherwise, if it is determined at the step 270 that the new bit allocation table is different than the old bit allocation table, then control transfers from the step 270 to a step 272 where a flag is sent from the receiver to the transmitter indicating that the bit allocation table has changed. In a preferred embodiment, the flag is sent at the step 272 via a single carrier of the multicarrier signal
15 that is reserved for use by the transmitter and receiver only for the flag. In another embodiment, the reserved carrier can also be used to transmit the new bit allocation table.

Following the step 272 is a step 274 where the receiver sends the new bit allocation table, determined at the step 268, to the transmitter. Following the step 274, control transfers back to the test step 262 to poll and wait for the transmitter to send another reference frame.
20

While the invention has been disclosed in connection with the preferred embodiments shown and described in detail, various modifications and improvements thereon will become readily apparent to those skilled in the art. Accordingly, the spirit and scope of the present invention is to be limited only by the following claims.

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CLAIMS

- 1 1. In a multicarrier modulation system having a plurality of channels for transmit-
2 ting data at varying rates from a transmitter to a receiver dependent on the signal to
3 noise ratio of the respective channels, the improvement comprising:
 - 4 A. means for allocating data to respective ones of said channels in accor-
5 dance with an initial signal to noise ratio for the corresponding channel,
 - 6 B. means for repetitively calculating trial noise margins across said channels,
7 and
 - 8 C. means for repetitively combining said trial noise margins with the said
9 signal to noise ratios of said channels to form modified signal to noise ra-
10 tios for said channels for use in reallocating said data thereto.
- 1 2. A multicarrier modulation system according to claim 1 in which said margins are
2 added to the constellation signal to noise ratios associated with said channels to form
3 said modified ratios.
- 1 3. A multicarrier modulation system according to claim 2 in which said margins are
2 added to said signal to noise ratios equally across said channels.
- 1 4. A multicarrier modulation system according to claim 3 which includes means for
2 defining upper and lower margin thresholds M_H and M_L , respectively, said trial noise
3 margins being defined as a combination of said thresholds.
- 1 5. A multicarrier modulation system according to claim 4 in which said combination
2 is formed as an average of said upper and lower thresholds.
- 1 6. A multicarrier modulation system according to claim 5 in which at least one of
2 said thresholds is determined as a function of the difference between the amount of data
3 transmissible across said channels in accordance with previously specified signal to noise
4 ratios associated with said channels and the amount of data desired to be transmitted
5 across said channels.

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- 1 7. A multicarrier modulation system according to claim 6 in which at least one of
2 said thresholds is set to zero.
- 1 8. A multicarrier modulation system according to claim 6 which includes means for
2 terminating data allocation when the amount of data transmissible across said channels in
3 accordance with previously specified signal to noise ratios associated with said channels
4 equals the amount of data desired to be transmitted across said channels.
- 1 9. A multicarrier modulation system according to claim 6 which includes means for
2 terminating data allocation when said difference diverges.
- 1 10. A multicarrier modulation system according to claim 6 which includes means for
2 terminating data allocation after a defined number of iterations of margin calculations
3 over said channels.
- 1 11. A multicarrier modulation system according to claim 1 in which said means for
2 calculating trial noise margins comprises:
- 3 A. means for defining a trial margin that is a function of the difference be-
4 tween the amount of data allocable to said channels in accordance with
5 said initial signal to noise ratios for the respective channels and the
6 amount of data desired to be transmitted, and
- 7 B. means for repetitively adjusting said trial margin in accordance with the
8 relation between the amount of data transmissible across said channels
9 when the signal to noise ratios of said channels are augmented by said
10 trial margin and the amount of data
- 11 C. transmissible across said channels in accordance with a prior determina-
12 tion of said signal to noise ratios.

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1 12. A multicarrier modulation system according to claim 1 which includes means for
2 periodically transmitting a reference frame from the transmitter to the receiver across said
3 channels, and means for measuring the signal to noise ratios of said channels from the
4 transmitted reference frame, said means for repetitively calculating trial noise margins
5 across said channels using the signal to noise ratios determined in the most recently
6 transmitted frame as the initial signal to noise ratios for calculating said margins in the in-
7 terval between said frame and the next frame.

1 13. A multicarrier modulation system according to claim 12 which includes first and
2 second memory register sets at both said transmitter and said receiver for storing channel
3 data allocations in accordance with signal to noise ratios associated therewith, and means
4 for transmitting from the transmitter to the receiver a flag indicating which of the register
5 sets is to be used for subsequently receiving data from said transmitter.

1 14. In a multicarrier modulation system having a plurality of channels for transmitting
2 data at varying rates from a transmitter to a receiver dependent on the signal to noise ratio
3 of the respective channels, the improvement comprising:

- 4 A. means for allocating data to respective ones of said channels in accordance
5 with initial signal to noise ratios measured for the corresponding channels,
- 6 B. means for calculating a trial noise margin across said channels as a function
7 of said initial signal to noise ratio and the difference between the amount of
8 data transmissible over said channels with said signal to noise ratios and
9 the amount of data desired to be transmitted,
- 10 C. means for augmenting the initial signal to noise ratios associated with said
11 channel by the trial noise margin to thereby define an augmented signal to
12 noise ratio for use in defining a revised estimate of the amount of data
13 transmissible over said channels,
- 14 D. means for repetitively defining successive trial noise margins as a function
15 of the augmented signal to noise ratios and the difference between the

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16 amount of data transmissible over said channels with said augmented signal
17 to noise ratios and the amount of data desired to be transmitted until an
18 exit condition is reached.

1 15. A multicarrier modulation system according to claim 14 in which said exit condi-
2 tion comprises equality between the amount of data transmissible over said channels with a
3 particular set of augmented signal to noise ratios and the amount of data desired to be
4 transmitted.

1 16. A multicarrier modulation system according to claim 14 in which said exit condi-
2 tion comprises an increase in the difference between the amount of data transmissible over
3 said channels with said augmented signal to noise ratios and the amount of data desired to
4 be transmitted as determined on successive calculations.

1 17. A multicarrier modulation system according to claim 14 in which said exit condi-
2 tion comprises determination of a defined number of successive trial noise margins.

1 18. A multicarrier modulation system according to claim 14 which includes means for
2 periodically transmitting a reference frame from the transmitter to the receiver across said
3 channels, and means for measuring the signal to noise ratios of said channels from the
4 transmitted reference frame, said means for calculating trial noise margins across said
5 channels using the signal to noise ratios determined in the most recently transmitted frame
6 as the initial signal to noise ratios for calculating said margins in the interval between said
7 frame and the next frame.

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1 19. A multicarrier modulation system according to claim 18 which includes first and
2 second memory register sets at both said transmitter and said receiver for storing chan-
3 nel data allocations in accordance with signal to noise ratios associated therewith, and
4 means for transmitting from the transmitter to the receiver a flag indicating which of the
5 register sets is to be used for subsequently receiving data from said transmitter.

1 20. A method of allocating data to respective ones of channels in a multicarrier
2 modulation system having a plurality of channels for transmitting data at varying rates
3 from a transmitter to a receiver, comprising the steps of :

- 4 A. allocating data to respective ones of said channels in accordance with
5 measured signal to noise ratio for the corresponding channel,
- 6 B. repetitively calculating trial noise margins across said channels, and
- 7 C. repetitively combining said trial noise margins with the said signal to
8 noise ratios of said channels to form modified signal to noise ratios for
9 said channels for use in reallocating said data thereto.

1 21. A method according to claim 20 in which the step of combining said trial noise
2 margins and said signal to noise ratios of said channels comprises adding a calculated
3 trial noise margin to the constellation signal to noise ratios of said channels to thereby
4 form an augmented signal to noise ratio from which the amount of data transmissible in
5 said channel is determined.

1 22. A method according to claim 20 in which the step of repetitively calculating trial
2 noise margins across said channels comprises the steps of

- 3 A. repetitively defining a trial margin that is a function of the difference be-
4 tween the amount of data allocable to said channels in accordance with
5 said initial signal to noise ratios for the respective channels and the
6 amount of data desired to be transmitted, and
- 7 B. repetitively adjusting said trial margin in accordance with the relation
8 between the amount of data transmissible across said channels when the
9 signal to noise ratios of said channels are augmented by said trial margin
10 and the amount of data transmissible across said channels in accordance
11 with a prior determination of said signal to noise ratios.

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- 1 23. A method according to claim 22 which further includes the steps of:
- 2 A. periodically transmitting a reference frame from the transmitter to the
- 3 receiver across said channels,
- 4 B. measuring the signal to noise ratios of said channels from the transmitted
- 5 reference frame and using the signal to noise ratios determined in the
- 6 most recently transmitted frame as the signal to noise ratios for calculat-
- 7 ing said margins in the interval between said frame and the next frame.
- 1 24. A method according to claim 23 which includes the steps of providing first and
- 2 second memory register sets at both said transmitter and said receiver for storing chan-
- 3 nel data allocations in accordance with signal to noise ratios associated therewith, and
- 4 transmitting from the transmitter to the receiver a flag indicating which of the register
- 5 sets is to be used for subsequently receiving data from said transmitter.

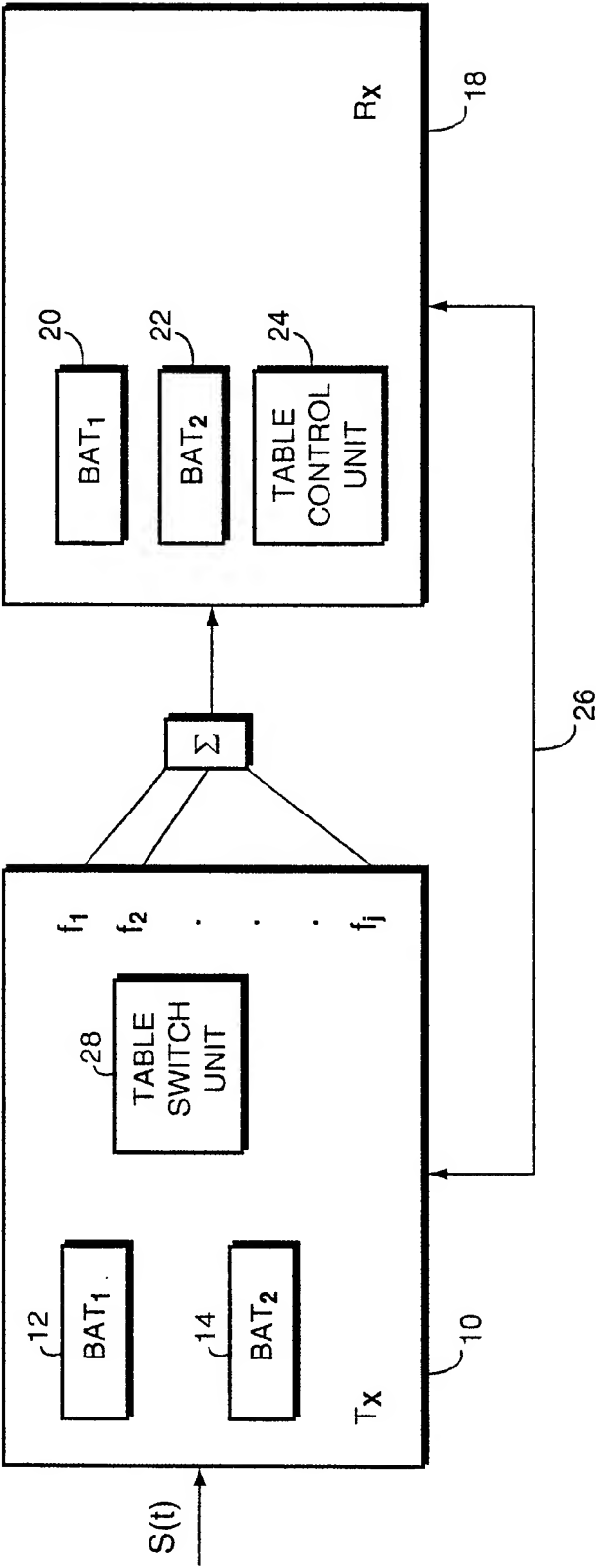


FIG. 1

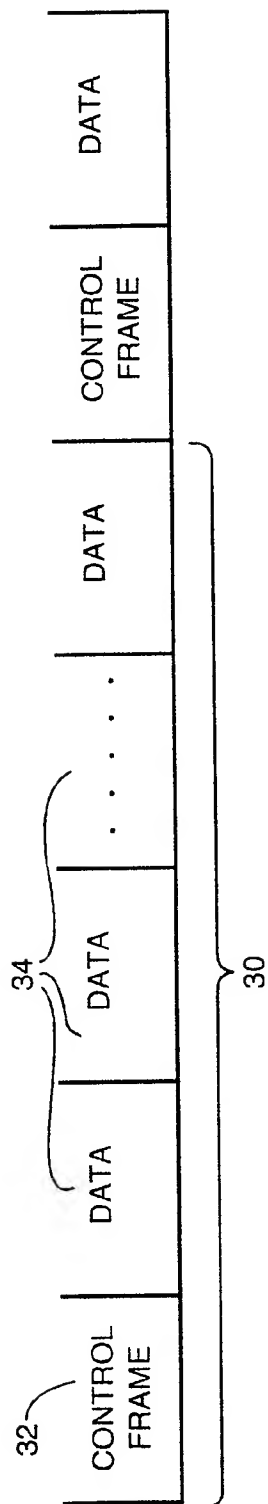


FIG. 2

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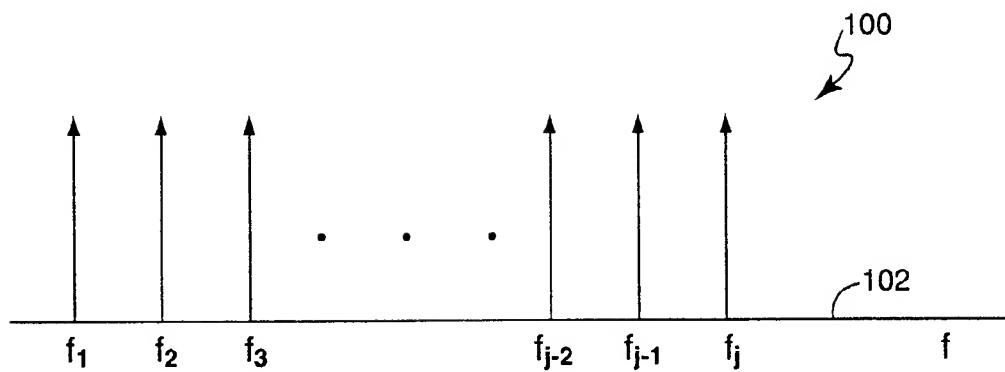


FIG. 3

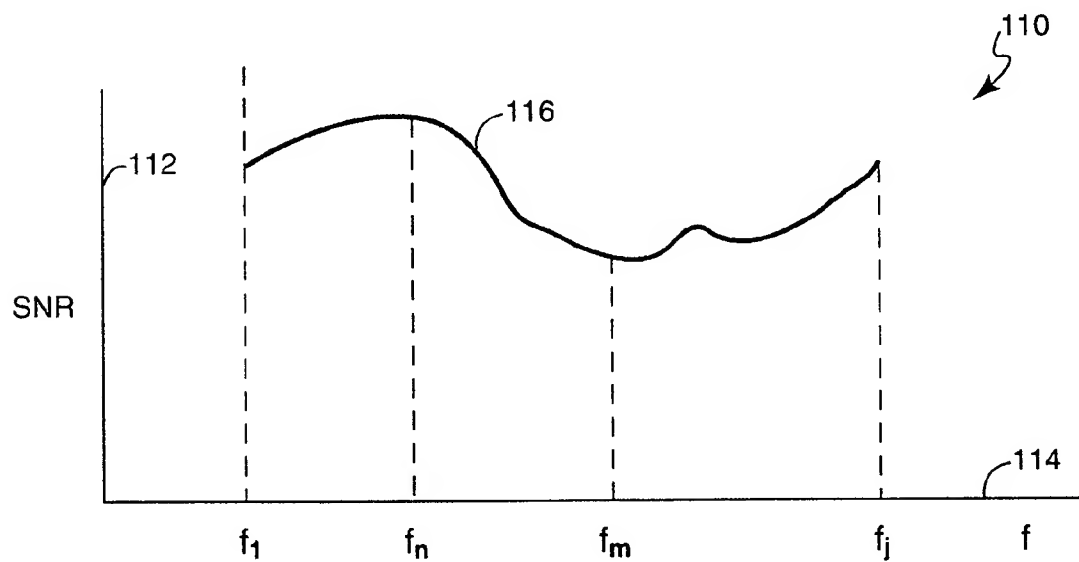


FIG. 4

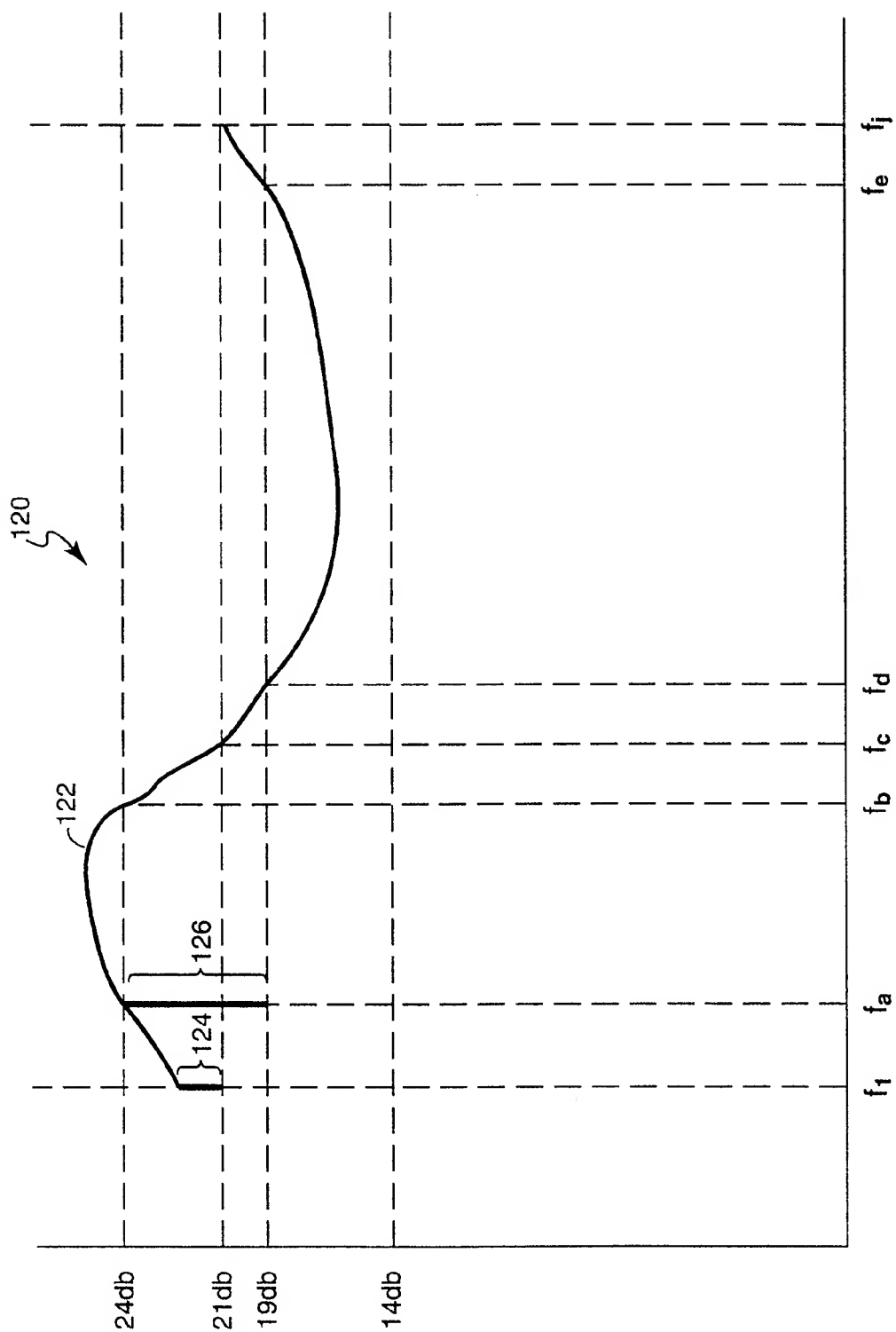


FIG. 5

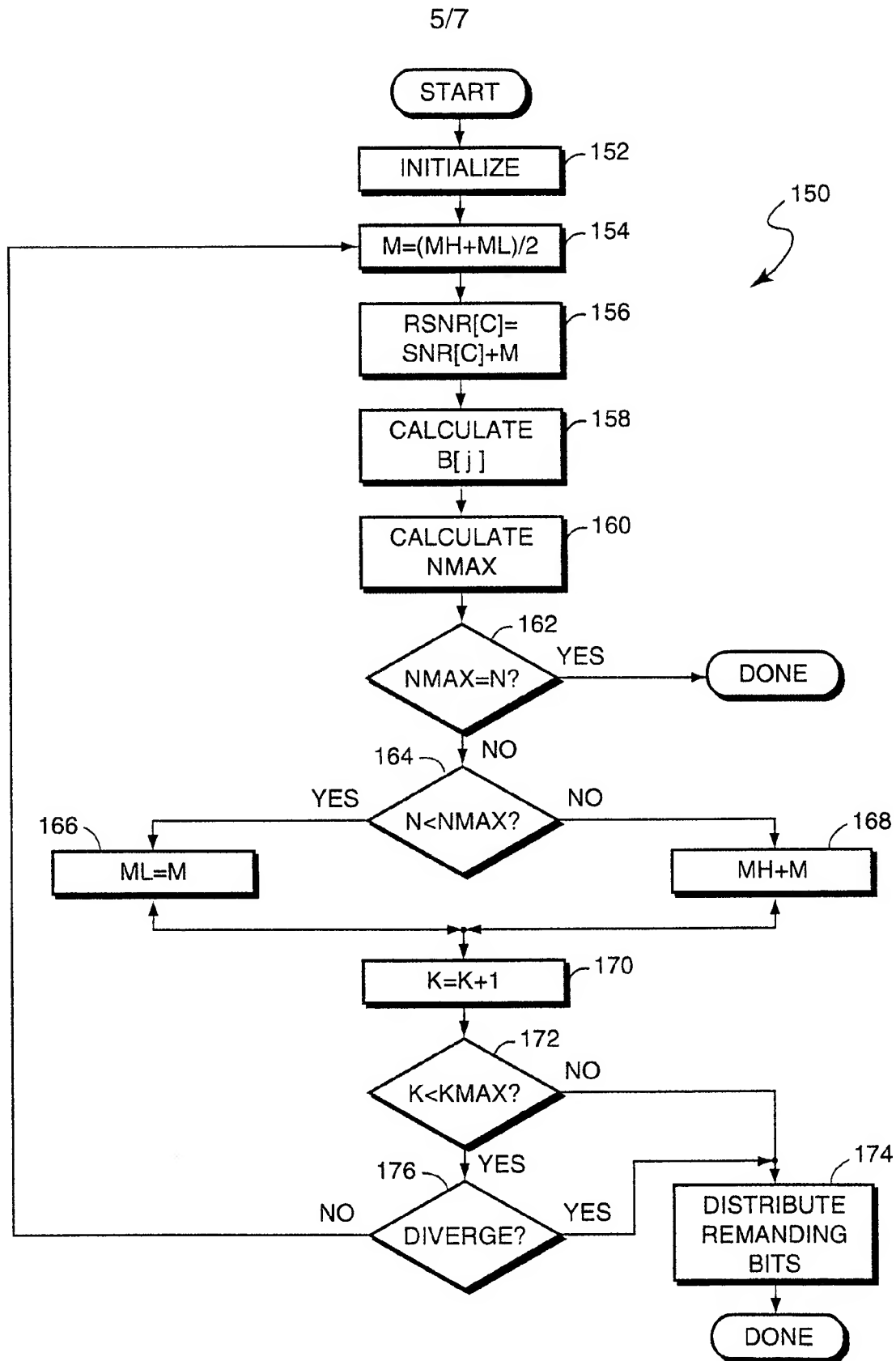
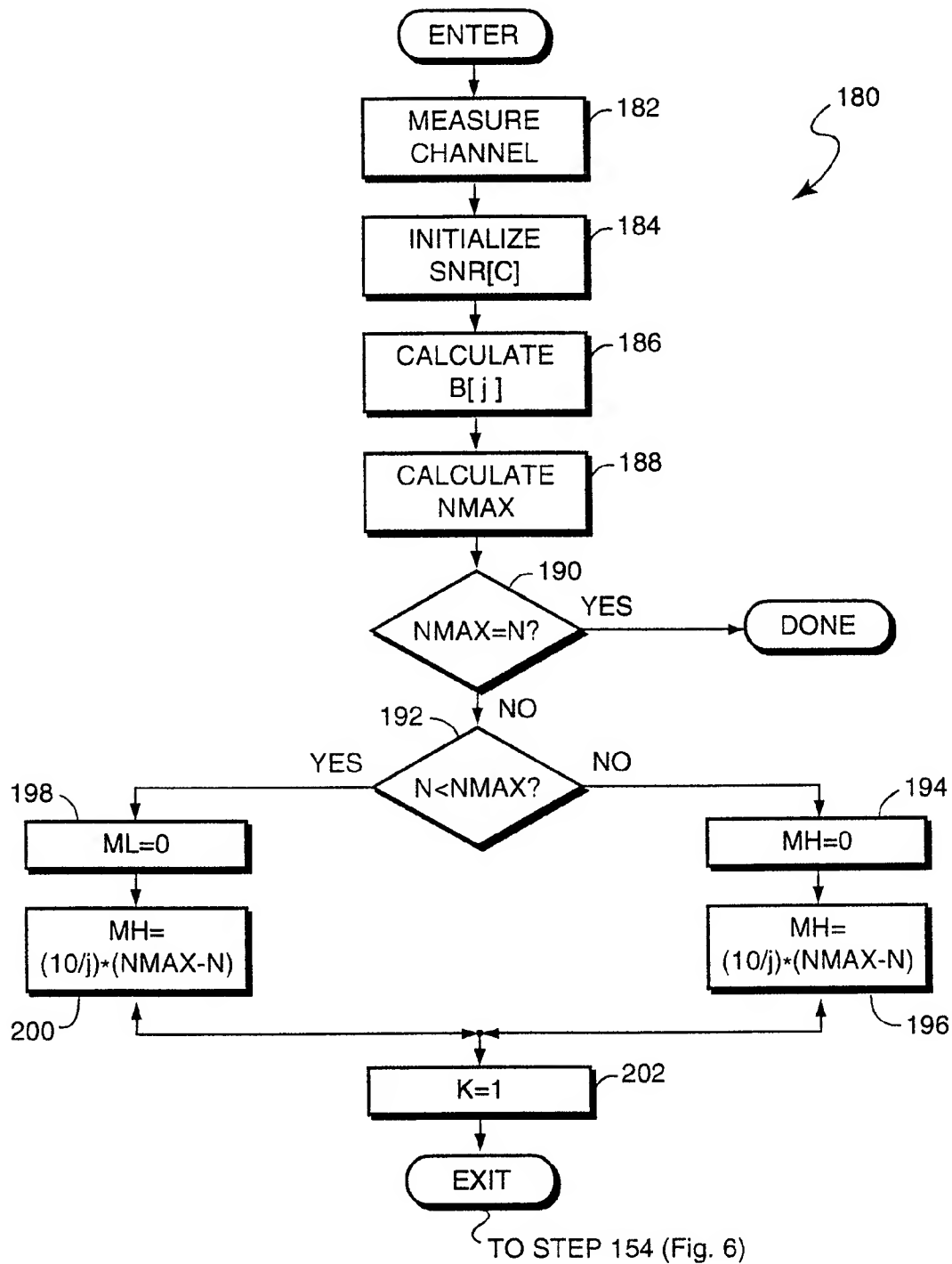
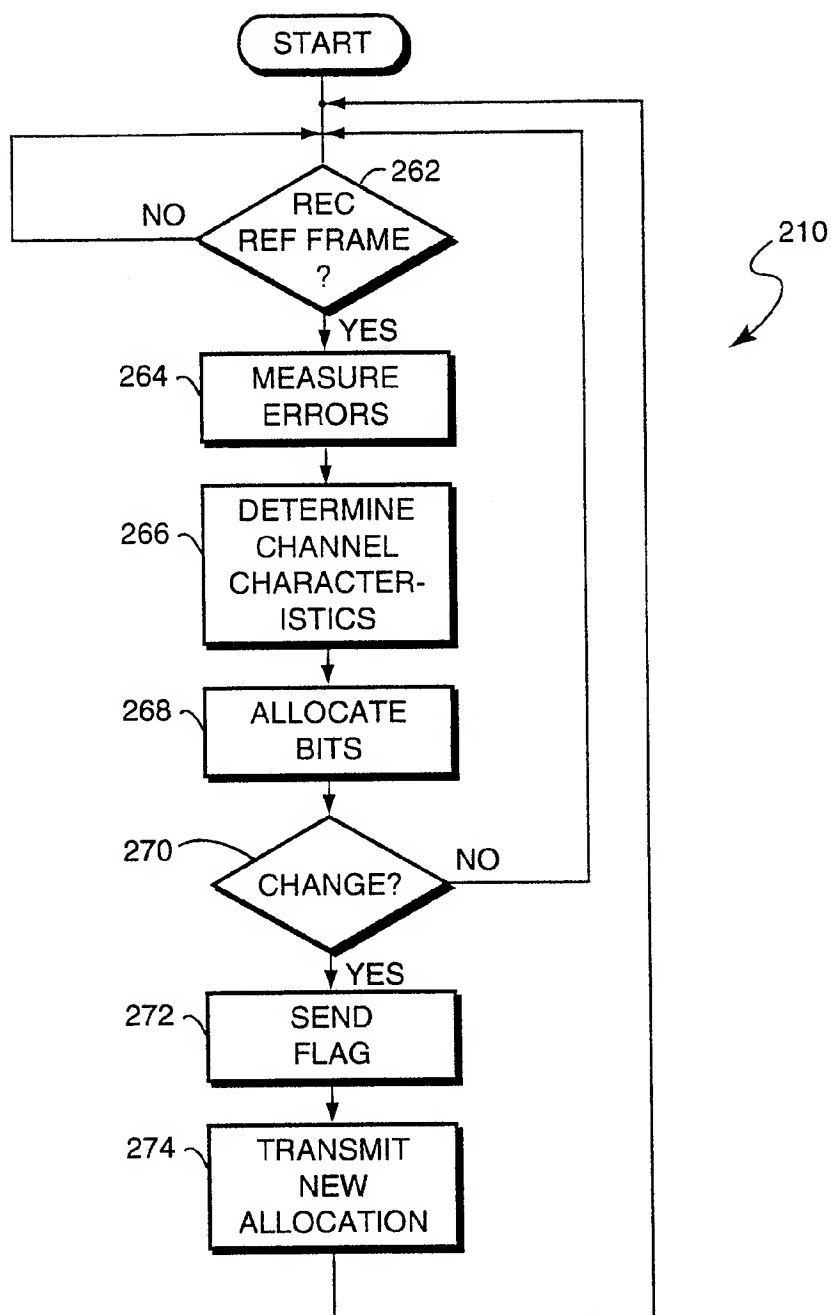


FIG. 6

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**FIG. 7**

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**FIG. 8**

INTERNATIONAL SEARCH REPORT

International Application No

PCT/US 98/11845

A. CLASSIFICATION OF SUBJECT MATTER

IPC 6 H04L27/26

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC 6 H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	EP 0 753 947 A (ALCATEL BELL NV) 15 January 1997 see column 6, line 39 - line 51 see column 11, line 49 - column 13, line 19 see claims 1-6,8 see figure 3 ---	1-24
A	US 5 479 447 A (CIOFFI JOHN M ET AL) 26 December 1995 cited in the application see column 3, line 57 - column 4, line 28 see column 5, line 42 - column 6, line 14 ---	1-24
A	WO 86 07223 A (TELEBIT CORP) 4 December 1986 cited in the application see page 17, line 12 - page 24, line 2 --- -/--	1-24

☒ Further documents are listed in the continuation of box C.

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Date of the actual completion of the international search

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INTERNATIONAL SEARCH REPORT

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C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
P,X	EP 0 812 087 A (MOTOROLA INC) 10 December 1997 see the whole document -----	1-24

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No

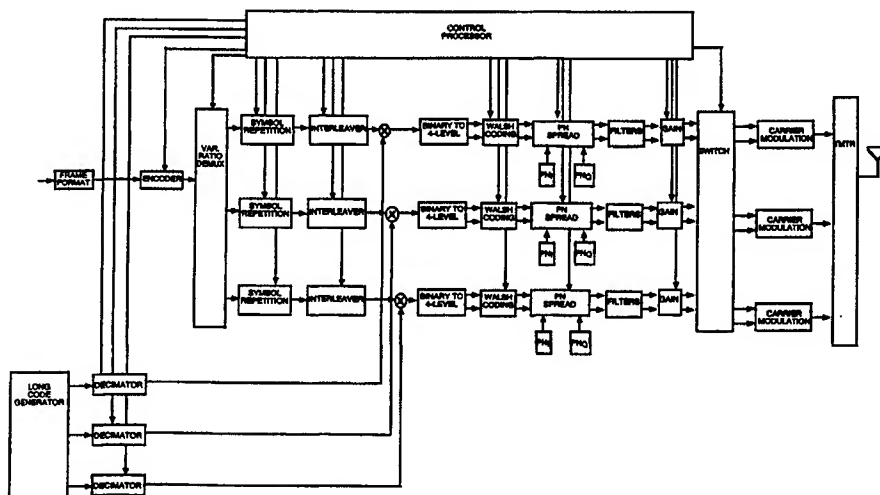
PCT/US 98/11845

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US 5479447 A	26-12-1995	NONE	
WO 8607223 A	04-12-1986	US 4679227 A AU 3528189 A AU 587037 B AU 5817786 A AU 609355 B AU 6009690 A BR 8606677 A CA 1251586 A CN 1011461 B CN 1048774 A CN 1048775 A,B CN 1048776 A,B DE 3681887 A DK 31087 A EP 0224556 A JP 6003956 B JP 62502932 T PT 82600 B US 4731816 A US 4833706 A US 5054034 A	07-07-1987 21-09-1989 03-08-1989 24-12-1986 26-04-1991 15-11-1990 11-08-1987 21-03-1989 30-01-1991 23-01-1991 23-01-1991 23-01-1991 14-11-1991 20-01-1987 10-06-1987 12-01-1994 19-11-1987 29-07-1994 15-03-1988 23-05-1989 01-10-1991
EP 0812087 A	10-12-1997	JP 10155031 A	09-06-1998

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(21) International Application Number: PCT/US98/19335 (22) International Filing Date: 16 September 1998 (16.09.98) (30) Priority Data: 08/931,536 16 September 1997 (16.09.97) US (71) Applicant: QUALCOMM INCORPORATED [US/US]; 6455 Lusk Boulevard, San Diego, CA 92121 (US). (72) Inventor: JOU, Yu-Cheun; 9979 Riverhead Drive, San Diego, CA 92129 (US). (74) Agents: MILLER, Russell, B. et al.; Qualcomm Incorporated, 6455 Lusk Boulevard, San Diego, CA 92121 (US).		(81) Designated States: AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, CA, CH, CN, CU, CZ, DE, DK, EE, ES, FI, GB, GE, GH, GM, HR, HU, ID, IL, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MD, MG, MK, MN, MW, MX, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, SL, TJ, TM, TR, TT, UA, UG, UZ, VN, YU, ZW, ARIPO patent (GH, GM, KE, LS, MW, SD, SZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GW, ML, MR, NE, SN, TD, TG). Published <i>With international search report.</i> <i>Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.</i>

(54) Title: A METHOD OF AND APPARATUS FOR TRANSMITTING DATA IN A MULTIPLE CARRIER SYSTEM

**(57) Abstract**

A method of an apparatus for transmitting data in a multiple carrier system comprises encoding data and dividing the resulting encoded symbols for transmission on different frequencies. The transmitter comprises a control processor (50) for determining the capacity of each of a plurality of channels and selecting a data rate for each channel depending on the determined capacity. A plurality of transmission subsystems (56 to 72) are responsive to the control processor (50). Each transmission subsystem is associated with a respective one of the plurality of channels for scrambling encoded data with codes unique to the channel for transmission in the channel. A variable demultiplexer (56) under the control of the control processor (50) demultiplexes the encoded data into the plurality of transmission subsystems at a demultiplexing rate derived from the data rates selected for the channels by the controller. In one embodiment of the transmission subsystems, the encoded symbols are provided to a symbol repetition unit (58) which keeps the symbol rate of data to be transmitted fixed. In another embodiment, no symbol repetition is provided and variable length Walsh sequences are used to handle data rate variations.

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A METHOD OF AND APPARATUS FOR TRANSMITTING DATA IN A MULTIPLE CARRIER SYSTEM

BACKGROUND OF THE INVENTION

5

I. Field of the Invention

The present invention relates to a method of and apparatus for transmitting data in a multiple carrier system. The present invention may be used for maximizing system throughput and increasing signal diversity by dynamically multiplexing signals onto multiple carriers in a spread spectrum communication system.

II. Description of the Related Art

15

It is desirable to be able to transmit data at rates which are higher than the maximum data rate of a single CDMA channel. A traditional CDMA channel (as standardized for cellular communication in the United States) is capable of carry digital data at a maximum rate of 9.6 bits per second using a 64 bit Walsh spreading function at 1.2288 MHz.

Many solutions to this problem have been proposed. One solution is to allocate multiple channels to the users and allow those users to transmit and receive data in parallel on the plurality of channels available to them. Two methods for providing multiple CDMA channels for use by a single user are described in co-pending U.S. Patent Application Serial No. 08/431,180, entitled "METHOD AND APPARATUS FOR PROVIDING VARIABLE RATE DATA IN A COMMUNICATIONS SYSTEM USING STATISTICAL MULTIPLEXING", filed April 28, 1997 and U.S. Patent Application Serial No. 08/838,240, entitled "METHOD AND APPARATUS FOR PROVIDING VARIABLE RATE DATA IN A COMMUNICATIONS SYSTEM USING NON-ORTHOGONAL OVERFLOW CHANNELS", filed April 16, 1997, both of which are assigned to the assignee of the present invention and are incorporated by reference herein. In addition, frequency diversity can be obtained by transmitting data over multiple spread spectrum channels that are separated from one another in frequency. A method and apparatus for redundantly transmitting data over multiple CDMA channels is described in U.S. Patent No. 5,166,951, entitled "HIGH CAPACITY SPREAD SPECTRUM CHANNEL", which is incorporated by reference herein.

The use of code division multiple access (CDMA) modulation techniques is one of several techniques for facilitating communications in which a large number of system users are present. Other multiple access communication system techniques, such as time division multiple access (TDMA), frequency division multiple access (FDMA) and AM modulation schemes such as amplitude companded single sideband (ACSSB) are known in the art. However, the spread spectrum modulation technique of CDMA has significant advantages over these other modulation techniques for multiple access communication systems.

The use of CDMA techniques in a multiple access communication system is disclosed in U.S. Patent No. 4,901,307, entitled "SPREAD SPECTRUM MULTIPLE ACCESS COMMUNICATION SYSTEM USING SATELLITE OR TERRESTRIAL REPEATERS", assigned to the assignee of the present invention and incorporated by reference herein. The use of CDMA techniques in a multiple access communication system is further disclosed in U.S. Patent No. 5,103,459, entitled "SYSTEM AND METHOD FOR GENERATING SIGNAL WAVEFORMS IN A CDMA CELLULAR TELEPHONE SYSTEM", assigned to the assignee of the present invention and incorporated by reference herein. Code division multiple access communications systems have been standardized in the United States in Telecommunications Industry Association Interim Standard IS-95, entitled "Mobile Station-Base Station Compatibility Standard for Dual Mode Wideband Spread Spectrum Cellular System", which is incorporated by reference herein.

The CDMA waveform by its inherent nature of being a wideband signal offers a form of frequency diversity by spreading the signal energy over a wide bandwidth. Therefore, frequency selective fading affects only a small part of the CDMA signal bandwidth. Space or path diversity on the forward/reverse link is obtained by providing multiple signal paths through simultaneous links to/from a mobile user through two or more antennas, cell sectors or cell-sites. Furthermore, path diversity may be obtained by exploiting the multipath environment through spread spectrum processing by allowing a signal arriving with different propagation delays to be received and processed separately. Examples of the utilization of path diversity are illustrated in co-pending U.S. Patent No. 5,101,501 entitled "SOFT HANDOFF IN A CDMA CELLULAR TELEPHONE SYSTEM", and U.S. Patent No. 5,109,390 entitled "DIVERSITY RECEIVER IN A CDMA CELLULAR TELEPHONE SYSTEM", both assigned to the assignee of the present invention and incorporated by reference herein.

FIG. 1 illustrates a transmission scheme for a multiple-carrier code division multiple access (CDMA) system in which each carrier carries a fixed fraction of the transmitted data. Variable rate frame of information bits are provided to encoder 2 which encodes the bits in accordance with a convolutional encoding format. The encoded symbols are provided to symbol repetition means 4. Symbol repetition means 4 repeats the encoded symbols so as to provide a fixed rate of symbols out of symbol repetition means 4, regardless of the rate of the information bits.

The repeated symbols are provided to block interleaver 6 which rearranges the sequence in which the symbols are to be transmitted. The interleaving process, coupled with the forward error correction, provides time diversity which aids in the reception and error recovery of the transmitted signal in the face of burst errors. The interleaved symbols are provided to data scrambler 12. Data scrambler 12 multiplies each interleaved symbol by +1 or -1 according to a pseudonoise (PN) sequence. The pseudonoise sequence is provided by passing a long PN sequence generated by long code generator 8 at the chip rate through decimator 10 which selectively provides a subset of the chips of the long code sequence at the rate of the interleaved symbol stream.

The data from data scrambler 12 is provided to demultiplexer (DEMUX) 14. Demultiplexer 14 divides the data stream into three equal sub-streams. The first sub-stream is provided to transmission subsystem 15a, the second sub-stream to transmission subsystem 15b and the third sub-stream to transmission subsystem 15c. The subframes are provided to serial-to-parallel converters (BINARY TO 4 LEVEL) 16a-16c. The outputs of serial to parallel converters 16a-16c are quaternary symbols (2bits/symbol) to be transmitted in a QPSK modulation format

The signals from serial-to-parallel converters 16a-16c are provided to Walsh coders 18a-18c. In Walsh coders 18a-18c, the signal from each converter 16a-16k is multiplied by a Walsh sequence consisting of ± 1 values. The Walsh coded data is provided to QPSK spreaders 20a-20c, which spread the data in accordance with two short PN sequences. The short PN sequence spread signals are provided to amplifiers 22a-22b which amplify the signals in accordance with a gain factor.

The system described above suffers from a plurality of drawbacks. First, because the data is to be provided in equal sub-streams on each of the carriers, the available numerology is limited to frames with a number of code symbols that will divide evenly by a factor of three. Table 1 below

illustrates the limited number of possible rate sets which are available using the transmission system illustrated in FIG. 1.

Walsh Function (QPSK Symbol) Rate [sps]	Number of Walsh Functions per 20ms		Length of Walsh Function [chips]	Symbol Rate [sps] (After Repetition)	Number of Symbols per 20 ms.	
1228800	24576	$3 \cdot (2^{13})$	1	2457600	49152	$3 \cdot (2^{14})$
614400	12288	$3 \cdot (2^{12})$	2	1228800	24576	$3 \cdot (2^{13})$
307200	6144	$3 \cdot (2^{11})$	4	614400	12288	$3 \cdot (2^{12})$
153600	3072	$3 \cdot (2^{10})$	8	307200	61444	$3 \cdot (2^{11})$
76800	1536	$3 \cdot (2^9)$	16	153600	3072	$3 \cdot (2^{10})$
38400	768	$3 \cdot (2^8)$	32	76800	1536	$3 \cdot (2^9)$
19200	384	$3 \cdot (2^7)$	64	38400	768	$3 \cdot (2^8)$
9600	192	$3 \cdot (2^6)$	128	19200	384	$3 \cdot (2^7)$
4800	96	$3 \cdot (2^5)$	256	9600	192	$3 \cdot (2^6)$
2400	48	$3 \cdot (2^4)$	512	4800	96	$3 \cdot (2^5)$
1200	24	$3 \cdot (2^3)$	1024	2400	48	$3 \cdot (2^4)$
600	12	$3 \cdot (2^2)$	2048	1200	24	$3 \cdot (2^3)$
300	6	$3 \cdot (2^1)$	4096	600	12	$3 \cdot (2^2)$
150	3	$3 \cdot (2^0)$	8192	300	6	$3 \cdot (2^1)$

5

Table 1

As illustrated in Table 1, because the symbols are evenly distributed to the three carriers, the total data rate is limited by the carrier with the least power available or requiring the highest SNR. That is the total data rate is equal to three times the data rate of the "worst" link (here the worst means the one requiring the highest SNR or having the least power available). this reduces the system throughput, because the worst link's rate is always chosen as the common rate for all three carriers, which results in under utilization of the channel resource on the two better links.

Second, frequency dependent fading can severely affect one of the frequencies while having a limited effect on the remaining frequencies. This implementation is inflexible and does not allow transmission of a frame to be provided in a way that reduces the effects of the poor channel. Third, because of frequency dependent fading, the fading will typically always affect the same groups of symbols of each frame. Fourth, were the

implementation to be superimposed on a speech transmission system there is no good way to balance the loads carried on the different frequencies on a frame by frame basis in the face of variable speech activities in each frame. This results in loss in total system throughput. And fifth, for a system with
5 only three frequency channels, with the implementation described, there is no method of separating the speech and data so as to provide the data on one frequency or set of frequencies and the speech on a different frequency or set of frequencies. This results in a loss of system throughput as mentioned above.

10 Therefore, there is a need felt for an improved multi-carrier CDMA communication system which offers greater flexibility in numerology and load balancing, better resolution in data rates supported, and which offers superior performance in the face of frequency dependent fading and uneven loading.

15

SUMMARY OF THE INVENTION

In one aspect the invention provides a transmitter for transmitting data at a data rate in a plurality of channels each having a capacity less than
20 the data rate, the transmitter comprising: a controller for determining the capacity of each of a plurality channels and selecting a data rate for each channel depending on the determined capacity; a plurality of transmission subsystems responsive to the controller and each associated with a respective one of the plurality of channels for scrambling encoded data with
25 codes unique to the channel for transmission in the channel; and a variable demultiplexer responsive to the controller for demultiplexing the encoded data into the plurality of transmission subsystems at a demultiplexing rate derived from the data rates selected for the channels by the controller.

In another aspect the invention provides a receiver comprising: a
30 receiving circuit for receiving signals simultaneously in a plurality of channels each of which signals define scrambled encoded symbols which together represent data from a common origin; a controller for determining a symbol rate for the signals in each channel; a plurality of receiving subsystems responsive to the controller and each associated with a
35 respective one of the plurality of channels for descrambling encoded symbols with codes unique to the channel to enable the data to be extracted therefrom; and a variable multiplexer responsive to the controller for multiplexing the data from the plurality of receiving subsystems at a

multiplexing rate derived from the symbol rates determined for the channels by the controller onto an output.

In a further aspect the invention provides a wireless transmitter, comprising: encoder for receive a set of information bits and encoding said
5 information bits to provide a set of code symbols; and a transmission subsystem for receiving said code symbols and for providing a subset of said code symbols on a first carrier frequency and the remaining symbols on at least one additional carrier frequency.

The invention also provides a method of transmitting data at a data
10 rate in a plurality of channels each having a capacity less than the data rate, the method comprising: determining the capacity of each of a plurality channels and selecting a data rate for each channel depending on the determined capacity; scrambling encoded data with codes unique to the channel for transmission in the channel; and demultiplexing the encoded
15 data into the plurality of channels at a demultiplexing rate derived from the data rates selected for the channels by the controller.

The invention further provides a method of receiving data, the method comprising: receiving signals simultaneously in a plurality of channels each of which signals define scrambled encoded symbols which
20 together represent data from a common origin; determining a symbol rate for the signals in each channel; descrambling encoded symbols in each channel with codes unique to the channel to enable the data to be extracted therefrom; and multiplexing the descrambled data from the plurality of channels at a multiplexing rate derived from the symbol rates determined
25 for the channels.

To better utilize the channel resource, it's necessary to be able to transmit a different data rate on each carrier according to the channel condition and the available power on each channel. One way of doing this is by changing the ratio of the inverse-multiplexing on to each of the carriers.
30 Instead of distributing the symbols with a ratio of 1:1:1, a more arbitrary ratio can be used together with different repetition schemes as long as the resulted symbol rate on each carrier is a factor of some Walsh function rate. Walsh function rate can be 1228800, 614400, 307200,..., 75 for Walsh function length from 1 to 16384.

35 Given the Walsh function length, if the symbol rate is lower than the Walsh function rate, symbol repetition is used to "match" the rate. The repetition factor can be any number, integer or fractional. It will be understood by one skilled in the art that when repetition is present, the total transmit power can be proportionately reduced to keep the code symbol

energy constant. The Walsh function length may or may not be the same on the three carriers, depending on whether we need to save code channels. For example, if the supportable code symbol rate on the three channels are 153600 sps, 30720 sps and 102400 sps (for rate 1/2 coding, these correspond to data rates of 76.8 kbps, 15.36 kbps and 51.2 kbps, respectively - the total data rate is 143.36 kbps), then the inverse-multiplexing ratio will be 15:3:10.

If a Walsh function of length 8 is used for all three channels (assuming QPSK modulation with a QPSK symbol rate of 153.6 Ksps), then each code symbol is transmitted twice, 10 times, and three times on the three channels, respectively. Additional time diversity can be obtained if the repeated symbols are further interleaved. In an alternative embodiment, different Walsh function lengths are used. For example, Walsh functions for the three channels in the example of above of length 16, 16 and 8 respectively can be used, with each code symbol transmitted once on the first channel, five times on the second, and three times on the third.

The above approach does not affect the encoder since it has to be able to handle the highest data rate anyway. All that is changed is the number of data octets at the encoder input. However, this approach does have an impact on the implementation of the interleaver because the interleaver will have many possible sizes (in terms of number of symbols) if all combinations of data rates on the three channels are allowed. One alternative to the above approach which mitigates this problem is to inverse-multiplex the code symbols out of the encoder to the three carriers directly and perform interleaving of repeated code symbols on each channel separately. This simplifies the numerology and reduces the number of possible interleaver sizes on each channel.

BRIEF DESCRIPTION OF THE DRAWINGS

Further features, objects, and advantages of the present invention will become more apparent from the detailed description set forth below of embodiments of the invention when taken in conjunction with the drawings in which like reference characters identify correspondingly throughout and wherein:

FIG. 1 is a block diagram illustrating a multiple frequency CDMA communication system with fixed rates and carriers;

FIG. 2 is a block diagram illustrating a transmission system embodying the present invention;

FIG. 3 is a block diagram illustrating a receiver system embodying the present invention; and

FIG. 4 is a table of code channel Walsh symbols in a traditional IS-95 CDMA communication system.

5

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Referring to FIG. 2, which is a block diagram illustrating a transmission system embodying the present invention, the first operation to be performed is to determine the amount of data which can be supported on each of the carriers. Three such carriers are illustrated in FIG. 2, though one skilled in the art will realize that the present invention is easily extended to any number of carriers. Control processor 50 based on a set of factors such as the loading on each of the carriers, the amount of data queued for transmission to the mobile station and the priority of the information to be transmitted to the mobile station determines the rate of data transmission on each of the carriers.

After having selected the data rate to be transmitted on each of the carriers, control processor 50 selects a modulation format that is capable of transmitting data at the selected rate. In the exemplary embodiment, different length Walsh sequences are used to modulate the data depending on the rate of the data to be transmitted. The use of different length Walsh sequences selected to modulate the data depending on the rate of the data to be transmitted is described in detail in co-pending U.S. Patent Application Serial No. 08/654,443, filed May 28, 1996, entitled "HIGH RATE DATA WIRELESS COMMUNICATION SYSTEM", which is assigned to the assignee of the present invention and incorporated by reference herein. In an alternative embodiment, the high rate data can be supported by bundling of CDMA channels as described in the aforementioned Patent Applications Serial Nos. 08/431,180 and 08/838,240.

Once the rates which will be supported on each of the carriers is selected then control processor 50 calculates an inverse multiplexing ratio that will determine the amount of each transmission that will be carried on each of the carriers. For example, if the supportable code symbol rate on the three channels are 153600 sps, 30720 sps and 102400 sps (for rate 1/2 coding, these correspond to data rates of 76.8 kbps, 15.36 kbps and 51.2 kbps, respectively - the total data rate is 143.36 kbps), then the inverse-multiplexing ratio will be 15:3:10.

In the exemplary embodiment, frames of information bits are provided to frame formatter 52. In the exemplary embodiment, formatter 52 generates and appends to the frame a set of cyclic redundancy check (CRC) bits. In addition, formatter 52 appends a predetermined set of tail bits. The
5 implementation and design of frame formatters are well known in the art, an example of a typical frame formatter is described in detail in U.S. Patent No. 5,600,754, entitled "METHOD AND SYSTEM FOR THE ARRANGEMENT OF VOCODER DATA FOR THE MASKING OF TRANSMISSION CHANNEL INDUCED ERRORS", which is assigned to the
10 assignee of the present invention and incorporated by reference herein.

The formatted data is provided to encoder 54. In the exemplary embodiment, encoder 54 is a convolutional encoder, though the present invention can be extended to other forms of encoding. A signal from control processor 50 indicates to encoder 54 the number of bits to be encoded
15 in this transmission cycle. In the exemplary embodiment, encoder 54 is a rate 1/4 convolutional encoder with a constraint length of 9. It should be noted that because of the additional flexibility provided by the present invention, essentially any encoding format can be used.

The encoded symbols from encoder 54 are provided to variable ratio de-multiplexer 56. Variable ratio de-multiplexer 56 provides the encoded symbols to a set of outputs based on a symbol output signal provided by control processor 50. In the exemplary embodiment, there are three carrier frequencies and control processor 50 provides a signal indicative of the number of encoded symbols to be provided on each of the three outputs. As
25 one skilled in the art will appreciate, the present invention is easily extended to an arbitrary number of frequencies.

The encoded symbols provided on each of the outputs of variable ratio de-multiplexer 56 are provided to a corresponding symbol repetition means 58a-58c. Symbol repetition means 58a-58c generate repeated versions
30 of the encoded symbols so that the resultant symbol rate matches with the rate of data supported on that carrier and the in particular matches Walsh function rate used on that carrier. The implementation of repetition generators 58a-58c is known in the art and an example of such is described in detail in U.S. Patent No. 5,629,955, entitled "Variable Response Filter",
35 which is assigned to the assignee of the present invention and incorporated by reference herein. Control processor 50 provides a separate signal to each repetition generator 58a-58c indicating the rate of symbols on each carrier or alternatively the amount of repetition to be provided on each carrier. In response to the signal from control processor 50, repetition means 58a-58c

generate the requisite numbers of repeated symbols to provide the designated symbol rates. It should be noted that in the preferred embodiment, the amount of repetition is not limited to integer number wherein all symbols are repeated the same number of times. A method for providing non-integer repetition is described in detail in co-pending U.S. Patent Application Serial No. 08/886,815, filed March 26, 1997, entitled "METHOD AND APPARATUS FOR TRANSMITTING HIGH SPEED DATA IN A SPREAD SPECTRUM COMMUNICATIONS SYSTEM", which is assigned to the assignee of the present invention and incorporated by reference herein.

The symbols from repetition generators 58a-58c are provided to a corresponding one of interleavers 60a-60c which reorders the repeated symbols in accordance with a predetermined interleaver format. Control processor 50 provides an interleaving format signal to each of interleavers 60a-60c which indicates one of a predetermined set of interleaving formats. In the exemplary embodiment, the interleaving format is selected from a predetermined set of bit reversal interleaving formats.

The reordered symbols from interleavers 60a-60c are provided to data scramblers 62a-62c. Each of data scramblers 62a-62c changes the sign of the data in accordance with a pseudonoise (PN) sequence. Each PN sequence is provided by passing a long PN code generated by long code or PN generator 82 at the chip rate through a decimator 84a-84c, which selectively provides ones of the spreading symbols to provide a PN sequence at a rate no higher than that provided by PN generator 82. Because the symbol rate on each carrier may be different from one another, the decimation rate of decimators 84a-84c may be different. Decimators 84a-84c are sample and hold circuits which sample the PN sequence out of PN generator 82 and continue to output that value for a predetermined period. The implementation of PN generator 82 and decimators 84a-84c are well known in the art and are described in detail in the aforementioned U.S. Patent No. 5,103,459. Data scramblers 62a-62c exclusively-OR the binary symbols from interleavers 60a-60c with the decimated pseudonoise binary sequences from decimators 84a-84c.

The binary scrambled symbol sequences are provided to serial to parallel converters (BINARY TO 4-LEVEL) 64a-64c. Two binary symbols provided to converters 64a-64c are mapped to a quaternary constellation with values ($\pm 1, \pm 1$). The constellation values are provided on two outputs from converters 64a-64c. The symbol streams from converters 64a-64c are separately provided to Walsh spreaders 66a-66c.

There are many methods of providing high speed data in a code division multiple access communication system. In the preferred embodiment, the Walsh sequence length is varied in accordance with the rate of the data to be modulated. Shorter Walsh sequences are used to modulate higher speed data and longer Walsh sequences are used to modulate lower rate data. For example, a 64 bit Walsh sequence can be used to transmit data at 19.2 Ksps. However, a 32 bit Walsh sequence can be used to modulate data at 38.4 Ksps.

A system describing variable length Walsh sequence modulation is described in detail in co-pending U.S. Patent Application Serial No. 08/724,281, entitled "HIGH DATA RATE SUPPLEMENTAL CHANNEL FOR CDMA TELECOMMUNICATIONS SYSTEM", filed January 15, 1997 and incorporated by reference herein. The length of the Walsh sequences used to modulate the data depend on the rate of the data to be transmitted. FIG. 4 illustrates the Walsh functions in a traditional IS-95 CDMA system.

In the preferred embodiment of the invention, the number of Walsh channels allocated for the high-rate data can be any value 2^N where $N = \{2, 3, 4, 5, 6\}$. The Walsh codes used by Walsh coders 66a-66c are $64/2^N$ symbols long, rather than the 64 symbols used with the IS-95 Walsh codes. In order for the high-rate channel to be orthogonal to the other code channels with 64-symbol Walsh codes, 2^N of the possible 64 quaternary-phase channels with 64-symbol Walsh are eliminated from use. Table I provides a list of the possible Walsh codes for each value of N and the corresponding sets of allocated 64-symbol Walsh codes.

N	Walsh _i	Allocated 64-Symbol Walsh Codes
2	+ , + , + , + , + , + , + , + , + , + , + , + , + , + + , - , + , - , + , - , + , - , + , - , + , - , + , - + , + , - , - , + , + , - , - , + , + , - , - , + , + , - , - + , - , - , + , + , - , - , + , + , - , - , + , + , - , - + , + , + , + , - , - , - , - , + , + , + , + , - , - , - , - + , - , + , - , - , + , - , + , + , - , - , + , - , - , + , - + , + , - , - , - , + , + , + , - , - , - , - , + , + , + , - , - , + , - , + , + , - , - , + , - , + , + , - + , + , + , + , + , + , + , - , - , - , - , - , - , - , - , - + , - , + , - , + , - , + , - , - , + , - , + , - , + , + , + , - , - , + , - , - , - , + , + , - , - , + , + , + , - , - , + , + , - , - , - , + , + , - , - , + , - + , - , - , + , + , - , - , - , - , - , - , + , + , + , + + , - , - , + , + , - , - , - , - , - , - , + , + , + , + + , - , - , + , + , - , - , + , - , + , - , + , - , + , - + , + , + , + , - , - , - , - , - , - , - , - , + , + , + , + + , - , + , - , - , + , - , + , + , - , + , + , - , + , - + , + , - , - , - , + , + , - , - , + , + , + , - , - + , - , - , + , - , + , + , - , - , + , - , + , - , + ,	0, 16, 32, 48 1, 17, 33, 49 2, 18, 34, 50 3, 19, 35, 51 4, 20, 36, 52 5, 21, 37, 53 6, 22, 38, 54 7, 23, 39, 55 8, 24, 40, 56 9, 25, 41, 57 10, 26, 42, 58 11, 27, 43, 59 12, 28, 44, 60 13, 29, 45, 61 14, 30, 46, 62 15, 31, 47, 63
3	+ , + , + , + , + , + , + , + + , - , + , - , + , - , + , - + , + , - , - , + , + , - , - + , - , - , + , + , - , - , + + , + , + , + , - , - , - , - + , - , + , - , - , + , - , + + , + , - , - , - , - , + , + + , - , - , + , - , + , + , -	0, 8, 16, 24, 32, 40, 48, 56 1, 9, 17, 25, 33, 41, 49, 57 2, 10, 18, 26, 34, 42, 50, 58 3, 11, 19, 27, 35, 43, 51, 59 4, 12, 20, 28, 36, 44, 52, 60 5, 13, 21, 29, 37, 45, 53, 61 6, 14, 22, 30, 38, 46, 54, 62 7, 15, 23, 31, 39, 47, 55, 63
4	+ , + , + , + + , - , + , - + , + , - , - + , - , - , +	0, 4, 8, ..., 60 1, 5, 9, ..., 61 2, 6, 10, ..., 62 3, 7, 11, ..., 63
5	+ , + + , -	0, 2, 4, ..., 62 1, 3, 5, ..., 63
6	+	0, 1, 2, ..., 63

Table I.

The + and – indicate a positive or negative integer value, where the preferred integer is 1. As is apparent, the number of Walsh symbols in each Walsh code varies as N varies, and in all instances is less than the number

of symbols in the IS-95 Walsh channel codes. Regardless of the length of the Walsh code, in the described embodiment of the invention the symbols are applied at a rate of 1.2288 Megachips per second (Mcps). Thus, shorter length Walsh codes are repeated more often. Control processor 50 provides a signal
5 to Walsh coding elements 66a-66c which indicates the Walsh sequence to be used to spread the data.

Alternative methods for transmitting high rate data in CDMA communication system also include methods generally referred to as channel bundling techniques. The present invention is equally applicable to
10 the channel bundling methods for providing high speed data in a CDMA communication system. One method of providing channel bundled data is to provide a plurality of Walsh channels for use by a signal user. This method is described in detail in the aforementioned U.S. Patent Application Serial No. 08/739,482. An alternative channel bundling technique is to
15 provide the user with use of one Walsh code channel but to differentiate the signals from one another by means of different scrambling signals as described in detail in co-pending U.S. Patent Application Serial No. 08/838,240.

The Walsh spread data is provided to PN spreaders 68a-68c, which
20 apply a short PN sequence spreading on the output signals. In the exemplary embodiment, the PN spreading is performed by means of a complex multiplication as described in detail in the aforementioned co-pending U.S. Patent Application Serial No. 08/784,281. Data channels D_I and D_Q are complex multiplied, as the first real and imaginary terms
25 respectively, with spreading codes PN_I and PN_Q , as the second real and imaginary terms respectively, yielding in-phase (or real) term X_I and quadrature-phase (or imaginary) term X_Q . Spreading codes PN_I and PN_Q are generated by spreading code generators 67 and 69. Spreading codes PN_I and PN_Q are applied at 1.2288 Mcps. Equation (1) illustrates the complex
30 multiplication performed.

$$(X_I + jX_Q) = (D_I + jD_Q)(PN_I + jPN_Q) \quad (1)$$

In-phase term X_I is then low-pass filtered to a 1.2288 MHz bandwidth
35 (not shown) and upconverted by multiplication with in-phase carrier $\cos(\omega_c t)$. Similarly, quadrature-phase term X_Q is low-pass filtered to a 1.2288 MHz bandwidth (not shown) and upconverted by multiplication with quadrature-phase carrier $\sin(\omega_c t)$. The upconverted X_I and X_Q terms are summed yielding forward link signal $s(t)$. The complex multiplication

allows quadrature-phase channel set to remain orthogonal to the in-phase channel set and therefore to be provided without adding additional interference to the other channels transmitted over the same path with perfect receiver phase recovery.

5 The PN spread data is, then, provided to filters **70a-70c** which spectrally shape the signals for transmission. The filtered signals are provided to gain multipliers **72a-72c**, which amplify the signals for each carrier. The gain factor is supplied to gain elements **72a-72c** by control processor **50**. In the exemplary embodiment, control processor **50** selects the
10 gain factor for each carrier in accordance with the channel condition and the rate of the information data to be transmitted on that carrier. As is known by one skilled in the art, data that is transmitted with repetition can be transmitted with lower symbol energy than data without repetition.

 The amplified signals are provided to an optional switch **74**. Switch
15 **74** provides the additional flexibility of channel hopping the data signals onto different carriers. Typically, switch **74** is only used when the number of carriers actually used to transmit the signal is smaller than the total number of possible carriers (3 in the present example).

 The data is passed by switch **74** to carrier modulators **76a-76c**. Each of
20 carrier modulators **76a-76c** upconvert the data to a different predetermined frequency. The upconverted signals are provided to transmitter **78** where they are combined with other similarly processed signals, filtered and amplified for transmission through antenna **80**. In the exemplary embodiment, the amplified frequency upon which each of the signals are
25 transmitted varies with time. This provides additional frequency diversity for the transmitted signals. For example a signal that is currently being transmitted through carrier modulator **76a** will at predetermined time interval be switched so as to be transmitted on a different frequency through carrier modulators **76b** or **76c**. In accordance with a signal from control
30 processor **50**, switch **74** directs an amplified input signal from gain multiplier **72a-72c** to an appropriate carrier modulator **76a-76c**.

 Turning to FIG. 3, a receiver system embodying the present invention is illustrated. The signal received at antenna **100** is passed to receiver (RCVR) **102**, which amplifies and filters the signal before providing it to
35 switch **104**. The data is provided through switch **104** to an appropriate carrier demodulator **106a-106c**. It will be understood by one skilled in the art that although the receiver structure is described for the reception of a signal transmitted on three frequencies, the present invention can easily be

extended to an arbitrary number of frequencies consecutive to one another or not.

When the carriers on which the data is transmitted are rotated or hopped to provide additional frequency diversity, switch 104 provides the received signal to a selected carrier demodulator 106a-106c in response to a control signal from control processor 125. When the carrier frequencies are not hopped or rotated, then switch 104 is unnecessary. Each of carrier demodulators 106a-106c Quaternary Phase Shift Keying (QPSK) demodulate the received signal to baseband using a different downconversion frequency to provide a separate I and Q baseband signals.

The downconverted signals from each of carrier demodulators 106a-106c are provided to a corresponding PN despreader 108a-108c which removes the short code spreading from the downconverted data. The I and Q signals are despread by complex multiplication with a pair of short PN code. The PN despread data is provided to Walsh demodulators 110a-110c, which uncover the data in accordance with the assigned code channel sequences. In the exemplary embodiment, Walsh functions are used in the generation and reception of the CDMA signals but other forms of code channel generation are equally applicable. Control processor 125 provides a signal to Walsh demodulators 110a-110c indicating the Walsh sequences to be used to uncover the data.

The Walsh despread symbols are provided to parallel-to-serial converters (4-LEVEL TO BINARY) 112a-112c, which map the 2-dimensional signal into a 1-dimensional signal. The symbols are then provided to descramblers 114a-114c. Descramblers 114a-114c descramble the data in accordance with a decimated long code sequence generated as described with respect to the decimated long code sequences used to scramble the data in FIG. 2.

The descrambled data is provided to de-interleavers (DE-INT) 116a-116c. De-interleavers 116a-116c reorder the symbols in accordance with selected de-interleaver formats that are provided by control processor 125. In the exemplary embodiment, control processor 125 provides a signal indicative of the size of the deinterleaver and the scheme of de-interleaving to each of de-interleavers 116a-116c. In the exemplary embodiment, the de-interleaving scheme is selected from a predetermined set of bit reversal de-interleaving schemes.

The de-interleaved symbols are then provided to symbol combiners 118a-118c which coherently combine those repeatedly transmitted symbols. The combined symbols (soft decisions) are then provided to variable ratio

5 multiplexer 120 which reassembles the data stream and provides the reassembled data stream to decoder 122. In the exemplary embodiment decoder 122 is a maximum likelihood decoder, the implementation of which is well known in the art. In the exemplary embodiment, decoder 122
10 contains a buffer (not shown) which waits until an entire frame of data has been provided to it before beginning the decoding process. The decoded frame is provided to CRC check means 124 which determines whether the CRC bits check and if so provides them to the user otherwise an erasure is declared.

10 Having thus described the invention by reference to a preferred embodiment it is to be well understood that the embodiment in question is exemplary only and that modifications and variations such as will occur to those possessed of appropriate knowledge and skills may be made without departure from the spirit and scope of the invention as set forth in the
15 appended claims and equivalents thereof.

I CLAIM:

CLAIMS

1. A transmitter for transmitting data at a data rate in a plurality
2 of channels each having a capacity less than the data rate, the transmitter
comprising;
4 a controller for determining the capacity of each of a plurality
channels and selecting a data rate for each channel depending on the
6 determined capacity;
a plurality of transmission subsystems responsive to the controller
8 and each associated with a respective one of the plurality of channels for
scrambling encoded data with codes unique to the channel for transmission
10 in the channel; and
a variable demultiplexer responsive to the controller for
12 demultiplexing the encoded data into the plurality of transmission
subsystems at a demultiplexing rate derived from the data rates selected for
14 the channels by the controller.
2. A transmitter as claimed in claim 1, further comprising an
2 encoder for generating the encoded data from frames of data input thereto.
3. A transmitter as claimed in claim 1 or 2, wherein each
2 transmission subsystem comprises a symbol repetition unit for repeating
symbols to output the same at a rate corresponding to the rate selected for
4 the channel by the controller.
4. A transmitter as claimed in claim 3, wherein each transmission
2 subsystem comprises an interleaving unit for reordering the repeated
symbols depending on an interleaving format determined by the controller.
5. A transmitter as claimed in claim 4, further comprising a long
2 code generator for generating a respective long code for each channel; and
in each transmission subsystem, a scrambler for scrambling the reordered
4 symbols using the respective code for the channel.
6. A transmitter as claimed in claim 5, wherein the long code
2 generator comprises for each channel a decimator unit for decimating a
generated long code at a decimation rate determined by the controller so as
4 to produce the respective long codes for each channel.

7. A transmitter as claimed in claim 6, further comprising
2 variable coding units in each transmission subsystem for modulating the
scrambled symbols from the scrambler.

8. A transmitter as claimed in claim 7, wherein the coding units
2 are arranged to modulate the scrambled symbols with a respective walsh
code.

9. A transmitter as claimed in claim 7 or 8, further comprising a
2 pseudo noise spreader in each channel for spreading the modulated
symbols.

10. A transmitter as claimed in any preceding claim, further
2 comprising:

a switch; and

4 a plurality of carrier modulators, wherein the switch is responsive to
the controller for switching the scrambled data from the plurality of
6 transmission subsystems between the plural carrier modulators for
modulation of the signals onto different carriers at different times.

11. A receiver comprising:

2 a receiving circuit for receiving signals simultaneously in a plurality
of channels each of which signals define scrambled encoded symbols which
4 together represent data from a common origin;

a controller for determining a symbol rate for the signals in each
6 channel;

a plurality of receiving subsystems responsive to the controller and
8 each associated with a respective one of the plurality of channels for
descrambling encoded symbols with codes unique to the channel to enable
10 the data to be extracted therefrom; and

a variable multiplexer responsive to the controller for multiplexing
12 the data from the plurality of receiving subsystems at a multiplexing rate
derived from the symbol rates determined for the channels by the controller
14 onto an output.

12. A receiver as claimed in claim 11, further comprising an
2 decoder for decoding the encoded data output from the multiplexer into
frames of data.

13. A receiver as claimed in claim 11 or 12, further comprising a
2 pseudo noise despreaders in each channel for despreding the scrambled
encoded symbols.

14. A receiver as claimed in claim 13, further comprising variable
2 decoding units in each receiving subsystem for demodulating the despread
symbols from the despreders.

15. A receiver as claimed in claim 14, wherein the decoding units
2 are arranged to demodulate the despread symbols with a respective walsh
code.

16. A receiver as claimed in claim 15, further comprising, in each
2 receiving subsystem, a descrambler for descrambling the despread symbols
using a respective long code for the channel.

17. A receiver as claimed in claim 16, wherein each receiving
2 subsystem comprises an deinterleaving unit for reordering the repeated
symbols depending on an interleaving format determined by the controller.

18. A receiver as claimed in claim 17, wherein each receiving
2 subsystem comprises a symbol combiner for combining symbols to output
the same to the demultiplexer at a rate corresponding to the rate determined
4 for the channel by the controller.

19. A receiver as claimed in any of claims 11 to 18, further
2 comprising:

a switch; and

4 a plurality of carrier demodulators, wherein the switch is responsive
to the controller for switching the received signals between the plural carrier
6 demodulators for demodulation of the signals into different receiving
subsystems at different times.

20. A wireless transmitter, comprising:

2 encoder for receive a set of information bits and encoding said
information bits to provide a set of code symbols; and

4 transmission subsystem for receiving said code symbols and for
providing a subset of said code symbols on a first carrier frequency and the
6 remaining symbols on at least one additional carrier frequency.

21. A method of transmitting data at a data rate in a plurality of
2 channels each having a capacity less than the data rate, the method
comprising;
4 determining the capacity of each of a plurality channels and selecting
a data rate for each channel depending on the determined capacity;
6 scrambling encoded data with codes unique to the channel for
transmission in the channel; and
8 demultiplexing the encoded data into the plurality of channels at a
demultiplexing rate derived from the data rates selected for the channels by
10 the controller.

22. A method as claimed in claim 21, further comprising an
2 encoder for generating the encoded data from frames of data input thereto.

23. A method as claimed in claim 21 or 22, further comprising
2 repeating symbols for each channel to output the same at a rate
corresponding to the rate selected for the channel.

24. A method as claimed in claim 23, further comprising
2 reordering the repeated symbols depending on a determined interleaving
format.

25. A method as claimed in claim 24, further comprising
2 generating a respective long code for each channel; and
scrambling the reordered symbols in each transmission subsystem
4 using the respective code for the channel.

26. A method as claimed in claim 25, wherein the long code is
2 generated for each by decimating a generated long code at a determined
decimation rate for each channel.

27. A method as claimed in claim 26, further comprising for
2 modulating the scrambled symbols with a code.

28. A method as claimed in claim 27, the scrambled symbols are
2 modulated with a respective walsh code.

29. A method as claimed in claim 27 or 28, further comprising
2 spreading the modulated symbols with pseudo noise.

30. A method as claimed in any of claims 21 to 29, further
2 comprising modulating the scrambled data onto different carriers at
different times.

31. A method of receiving data, the method comprising:
2 receiving signals simultaneously in a plurality of channels each of
which signals define scrambled encoded symbols which together represent
4 data from a common origin;
determining a symbol rate for the signals in each channel;
6 descrambling encoded symbols in each channel with codes unique to
the channel to enable the data to be extracted therefrom; and
8 multiplexing the descrambled data from the plurality of channels at a
multiplexing rate derived from the symbol rates determined for the
10 channels.

32. A method of receiving data as claimed in claim 31, further
2 comprising decoding the multiplexed encoded data into frames of data.

33. A method of receiving data as claimed in claim 31 or 32, further
2 comprising despread the scrambled encoded symbols using a pseudo
noise code.

34. A method of receiving data as claimed in claim 33, further
2 comprising demodulating the despread symbols by way of variable decoding.

35. A method of receiving data as claimed in claim 34, wherein the
2 despread symbols are demodulated with a respective walsh code.

36. A method of receiving data as claimed in claim 35, further
2 comprising descrambling the despread symbols in each channel using a
respective long code for the channel.

37. A method of receiving data as claimed in claim 36, further
2 comprising reordering the repeated symbols depending on a determined
interleaving format.

38. A method of receiving data as claimed in claim 37, further
2 comprising combining symbols in a channel before demultiplexing the same
at a rate corresponding to the rate determined for the channel.

39. A method of receiving data as claimed in any of claims 31 to 38,
2 further comprising:
demodulating the signals in different channels at different times.

4

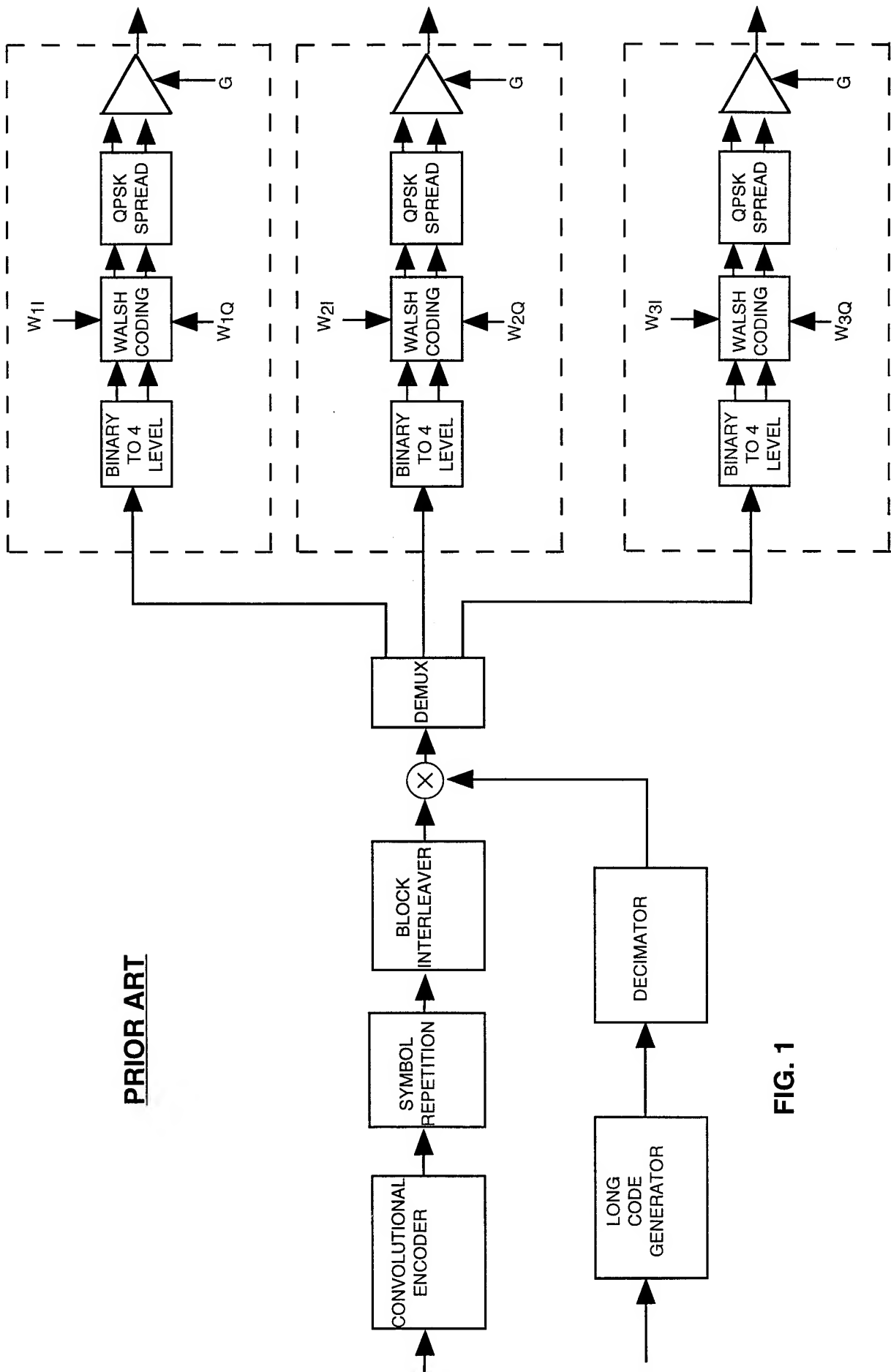


FIG. 1

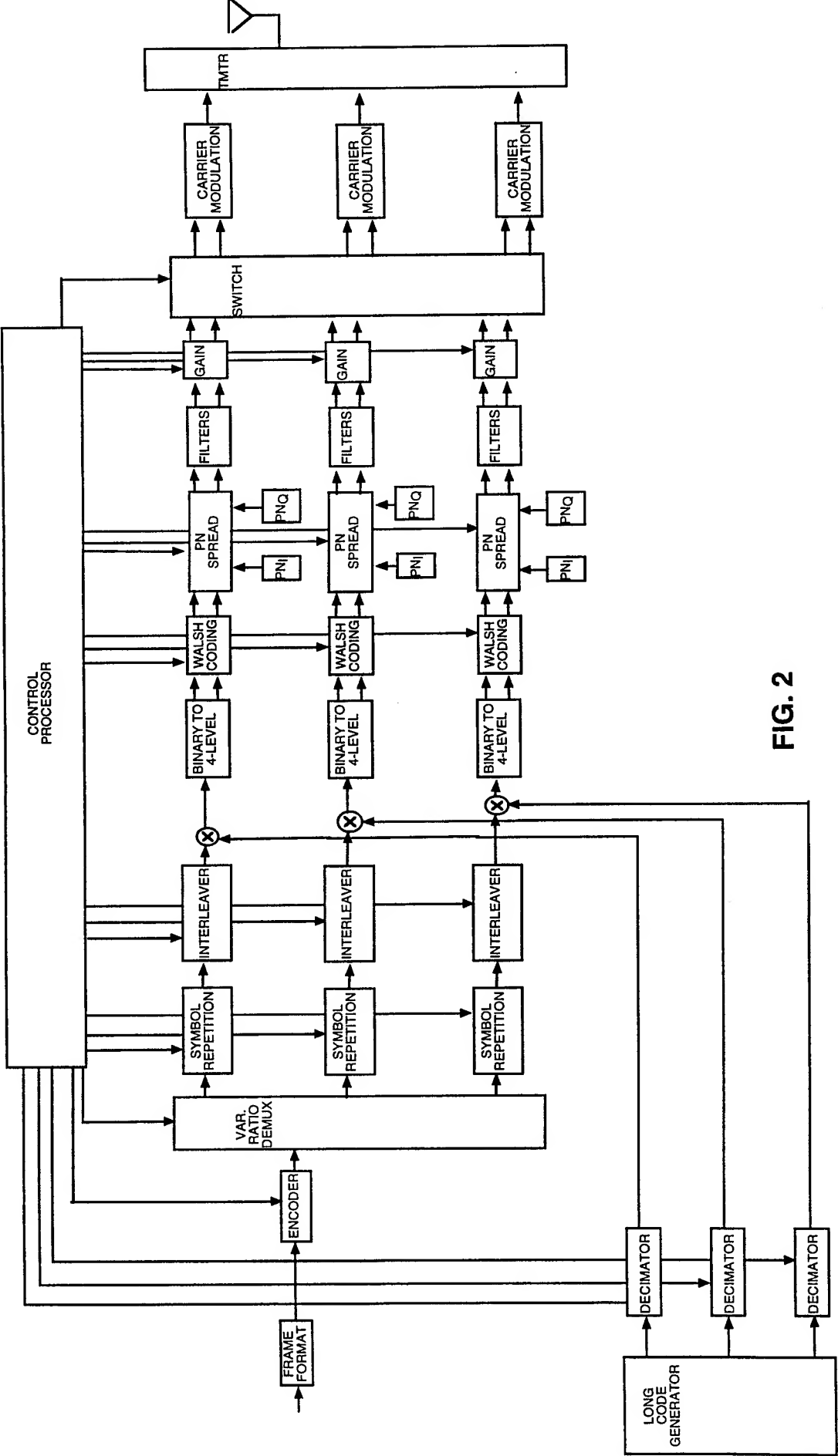


FIG. 2

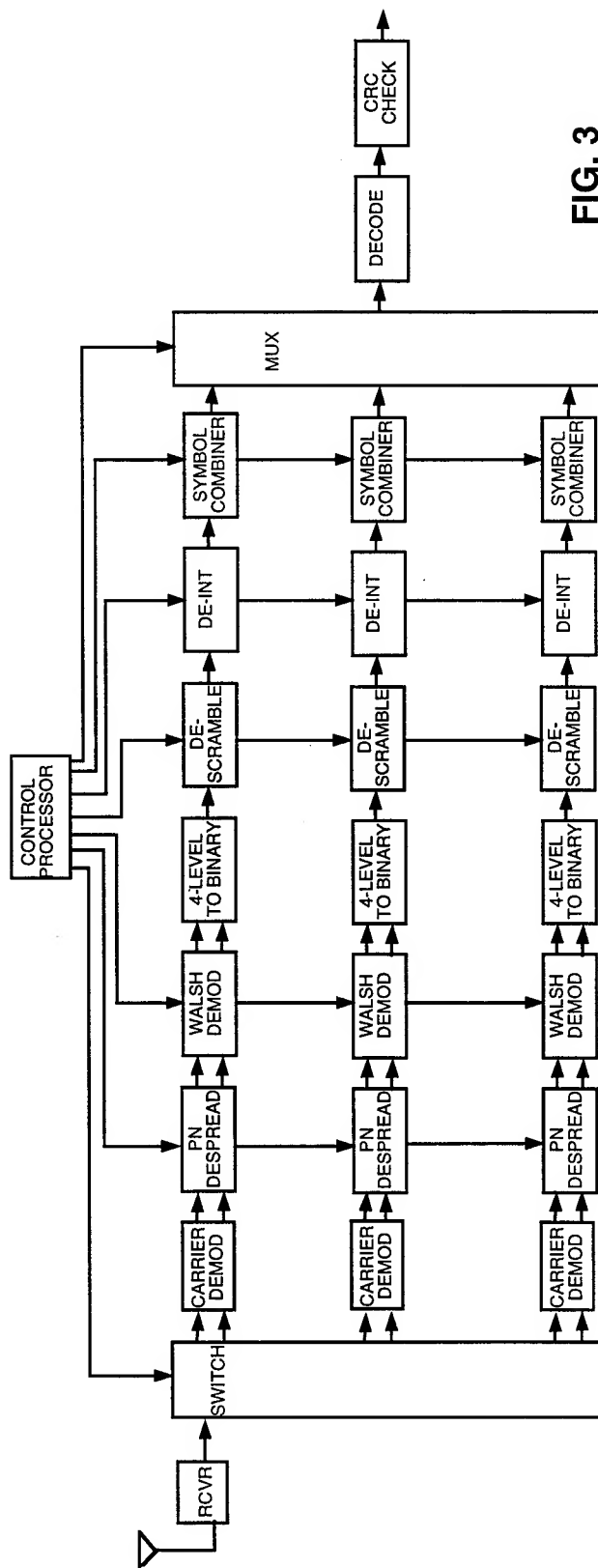


FIG. 3

